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FSVS-220
REVISION C

FSVS TERMINAL PERFORMANCE SPECIFICATION

Prepared For:

National Security Agency
9800 Savage Road
Fort George G. Meade, MD 20755-6000

Prepared By:

GTE

Electronic Defense Communications Directorate
GTE Government Systems Corporation
100 First Avenue
Waltham, MA 02154

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3.7 ELECTROSTATIC DISCHARGE (ESD) (MER)

The Type I and Type II terminals shall be designed to withstand ESD pulses encountered in their operating environment with no damage to the terminal and no loss of key. These pulses would be discharged into and out of the various user-accessible surfaces (e.g., keyboard, handset, keyceptacle) of the terminal. This protection shall be provided for a charged user touching either the terminal or a KSD-64A which has been inserted into the equipment, or for a charged KSD-64A which touches the terminal.

Table of Contents

3. The Magnetic Induction Test shall be performed using spikes and 60 Hz power frequency signals.
4. The radiated susceptibility signal applied to the terminal shall be modulated with signals that analysis indicates should have the greatest effect on the terminal.

3.4 HIGH-ALTITUDE ELECTROMAGNETIC PULSE (HEMP) CAPABILITY

The Government may have a number of users of the terminal who require protection against a free field/high altitude EMP threat. If the STU-III LCT is required to have an optional HEMP-hardened configuration, it shall be hardened to a level that satisfies interface characteristics of Specification NSA No. 77-27C, Appendix C, paragraphs 20.3.4.1 and 20.3.4.2. It is recommended that all STU-III LCT designs consider the requirements of paragraph 20.3.4.3 for typical telephone equipment. This requirement could be satisfied by employing an optional surge arrestor or device mounted externally to the terminal. It is not intended that all Type I terminals be burdened with an optional design feature for a limited number of users.

3.5 SAFETY REQUIREMENTS (MER)

The safety of the terminal shall be in accordance with MIL-STD-454, Requirement 1; and Underwriter's Laboratories requirements.

3.6 INSTALLATION (MER)

The terminal shall be easy to install. The goal shall be that an untrained user can install his own keyed terminal into a standard TELCO plug using at most one page of typewritten instructions.

Table of Contents (Cont.)

Characteristic	MIL-STD-461B/462	
Powerline Conducted Emissions	(note 1)	CE01 CE03
Radiated Electric Field Emissions		RE02
Radiated Magnetic Field Emissions		RE01
Powerline Conducted Susceptibility	(note 2) (note 2)	CS01 CS02 CS06
Magnetic Field Susceptibility	(note 3)	RS01
Radiated Susceptibility, Electric Field	(note 4)	RS03
Radiated Susceptibility, Magnetic Induction Field	(note 4)	RS02

Notes:

1. The limit for Powerline Conducted Emissions depends upon the current drain of the equipment and will be established after completion of the detailed equipment design.
2. The susceptibility signal applied to the powerline shall be modulated with signals that analysis indicates should have the greatest effect on the terminal.

Table of Contents (Cont.)

3.2.2.1 Overload Protection

It is recommended that the STU-III provide electrical overload protection which should conform to requirement 8 of MIL-STD-454 for class 1 equipment.

3.2.2.2 Leakage Current

It is recommended that the RMS leakage current caused by the electrical filters connected from phase to ground be limited to 30mA per phase for each unloaded input to the equipment as defined in paragraph 3.5.5 of MIL-E-16400.

3.3 EMI/EMC (MER)

3.3.1 EMI/EMC Requirements

The STU-III shall be designed to comply with FCC Rules, Part 15, Subpart J, Class B for radiated emissions via the electric field and for power line conducted emissions. The STU-III shall meet an electric field requirement of 100 μV/M from 30 MHz to 88 MHz, 150 μV/M from 88 MHz to 216 MHz and 200 μV/M from 216 MHz to 1000 MHz (all measured at three meters from the equipment). The STU-III shall meet a conducted emissions requirement of 250 μV from 450 KHz to 30 MHz. The STU-III terminals shall also meet the radiated susceptibility test specified by RS03 of MIL-STD-461B using the test method of MIL-STD- 462 with a frequency range from 450 KHz to 1 GHz at a field intensity of one volt/meter. The EMI/EMC design shall not degrade the ability of the terminal to comply with TEMPEST requirements of 4.1.

3.3.2 EMI/EMC Recommendations

The STU-III should comply with the requirements of MIL-STD-461B for Class A3 equipment and Part 15 of FCC Rules and Regulations in all modes of operation. The following characteristics should be verified by testing:

Table of Contents (Cont.)

3.2 POWER

3.2.1 Power Requirements

3.2.1.1 CONUS (MER)

The STU-III shall interface from a power source of 115V AC ± 10 percent, 60Hz ± 1 Hz via a three-prong plug or from an external DC power source selected by the contractor.

3.2.1.2 Foreign Country

The power supply for those STU-III terminals which are configured for operation in foreign countries shall be capable of operating from power sources of 91 to 134 VAC and 186 to 252 VAC, 47-63 Hz, single phase, non-polarity sensitive.

3.2.1.3 Leakage Current (MER)

The RMS leakage current caused by electrical filters connected from phase to ground shall not exceed 5 mA per phase for each unloaded input to the equipment, as defined in paragraph 3.11.9 of MIL-E-16400. Although MIL-E-16400 allows leakage currents in excess of 5 mA if warning labels are placed on the front panel of the equipment, the requirement for the STU-III LCT shall be a maximum 5 mA unless specifically waived by the Government.

3.2.2 Power Recommendations

The STU-III should provide a capability for normal telephone service, i.e., POTS mode, when prime power is lost. This capability may be integral to the terminal or it may be offered as an optional module.

Table of Contents (Cont.)

3.1.2.4 Flexibility/Modularity

A terminal that can be changed to meet future telecommunications requirements or take advantage of future technology breakthroughs is highly desirable. The scope of these changes could include speech processor or modem improvements, new key generator or key management techniques, new commercial offerings, and meeting a variety of existing or new telecommunications networks and protocols. It appears that the most attractive way of achieving flexibility is through a modular terminal design.

3.1.2.5 Form Factor

Versatility of the terminal would be enhanced with the capability for:

a. Transport in a standard briefcase.
b. Storage in a standard safe drawer.
c. Conversion to a rack mount.

3.1.2.6 Additional Office Environment Features

For additional applications of the LCT in the modern office environment, it would be useful for the terminal to provide modern office telecommunications capabilities currently available (e.g., call transfer, camp-on, intercom and multiparty interfaces, enabling/disabling individual phones from incoming ring signals, etc.). Operation of multiple desksets homed off one or more terminals or multiple terminals homed off of one or more PABX/CENTREX lines are also examples of useful features to be considered. Multiple synchronizations or FIREFLY rekeying actions are to be avoided. These capabilities may be integrated into the terminal or may be offered as an optional module.

3.1.2.7 Underwriters Laboratory

It is recommended that the STU-III be listed with Underwriters Laboratory, Inc.

Table of Contents (Cont.)

NOTE: The above information shall be on all terminals, however, manufacturers are free to add other information for their own purposes of control, warranty, modifications, place and date of manufacture, etc. This additional marking shall not interfere with the Government marking.

3.1.1.3 EIA RS-449/RS-232-C Interface Capability

The STU-III LCT shall provide an interface connector appropriate to support the RS-449 or RS-232-C interface provided with the terminal.

3.1.1.4 Interconnect

The STU-III LCT shall provide a permissive jack (RJ11C jack compatible) to interconnect with Telco and PABX facilities.

3.1.2 Physical Recommendations.

3.1.2.1 Modular Design of the BLACK I/O

A modular design would provide versatility to satisfy customer requirements today and leave the capability for interface to future systems at minimal cost.

3.1.2.2 Abbreviated Dialing

Abbreviated dialing should be considered if obtainable at a low cost.

3.1.2.3 Interconnect

The STU-III should provide interconnect with TELCO and PABX facilities via a programmable (RS45S) approach.

List of Illustrations

List of Tables

SECTION 6
ERRATA

To be Supplied Later.

SECTION 1
INTRODUCTION

This document specifies all of the minimum essential performance requirements and provides recommendations for additional features for the Type I and Type II STU-III Low Cost Terminals (LCTs) of the Future Secure Voice System (FSVS). It documents the functional and design characteristics, security design requirements and identifies recommendations for the Type I and Type II terminal configurations. This document also includes data patterns that can be used to verify the performance of critical functions. It is intended that this specification, along with the FSVS Signaling Plan-Interoperable Modes (FSVS Document No. FSVS-210), fully define all of the Government requirements for the STU-III/LCT.

This specification is not intended to force selection of components or technologies, nor force specific implementations. Specific components, technologies, and implementations which may be described are solely for the purpose of clarity and should not be considered a design requirement.

1.1 GENERAL PROGRAM OBJECTIVES

The overall objective of the FSVS program is to produce reliable, low cost, user-friendly secure voice and data equipment for large communities of subscribers. The system will be capable of serving as a minimum:

- 200,000 subscribers requiring protection for unclassified information and classified information at levels up to Top Secret, SI.
- 300,000 subscribers requiring protection for Unclassified but National Security Related (NSR) information.
- 600,000 Unclassified users requiring privacy for their communications.

The FSVS will include a family of subscriber terminals (with a STU-III designation) including the Low Cost Terminals (LCTs) to satisfy these system objectives. Throughout this specification, the terms STU-III, LCT, and STU-III/LCT are synonymous and should be interpreted as referring to the STU-III/LCT unless otherwise stated.

1.2 GENERAL TERMINAL REQUIREMENTS (MER)

All FSVS STU-III family terminals shall be interoperable in both a non-secure (or Clear) mode and in a common secure mode if properly keyed. In the secure mode, the STU-III/LCT shall provide terminal authentication and appropriate communications security for all levels of classified traffic passed.

The STU-III/LCT design shall provide acceptable speech quality and modem performance when operating over a variety of wireline networks. It shall also be capable of interoperation with terminals connected to a mobile (e.g., Cellular radio) network. The LCTs shall be simple to install, simple to operate by an untrained user, and suitable for an office environment.

1.3 TERMINAL CONFIGURATIONS (MER)

There shall be two distinct configuration types of LCT models:

- Type I - Capable of supporting all levels of classification of traffic.
- Type II - Capable of supporting Unclassified-NSR and other unclassified (e.g., public sector privacy) traffic.

The Type I terminal shall be a Controlled COMSEC Item (CCI) equipment. The Type II terminal, shall not be a Controlled COMSEC equipment but shall be accountable until point of sale. Both the Type I and Type II terminal shall be capable of being disabled from secure communications by removal of the Crypto Ignition Key (CIK).

MIL-STD-454J, "Standard General Requirements for Electronic Equipment", dtd 30 August 1984

MIL-STD-461B, "Electromagnetic Emission and Susceptibility Requirements for the Control of Electromagnetic Interface", dtd 1 April 1980

MIL-STD-462, "Electromagnetic Interface Characteristics - Measurement of", dtd 31 July 1967

MIL-STD-1189, "Bar Code Symbology Standard - DoD", dtd 4 September 1984

MIL-STD-1472C, "Human Engineering Design Criteria for Military Systems, Equipment and Facility", dtd 2 May 1981

NSTISSAM Tempest/1-91, "Compromising Emanations Laboratory Test Requirements, Electromagnetics", (Level 1), 21 March 1991

NSA No.77-27C, Classified Document and Title, Appendix C, dtd 6 October 1978

1.4 ORGANIZATION OF THE SPECIFICATION

This document provides a performance specification for both the Type I and Type II STU-III/LCT equipment. The document identifies the Minimum Essential Requirements (MERs) for each terminal type. These are firm requirements that must be incorporated into, or otherwise satisfied by any LCT design. In addition, this document also identifies a number of design recommendations that the Government feels will enhance the terminal/system utility, marketability, and versatility if they were available. These recommendations are clearly identified in the text. It is not mandatory for a STU-III/LCT design to incorporate these recommended features.

The remainder of this specification is divided into four Chapters. Chapter 2.0 presents the detailed functional and performance requirements and recommendations for the Type I and Type II STU-III/LCT terminals, and Chapter 3.0 includes specific design characteristics for both types. Chapter 4.0 includes requirements and recommendations for all security aspects of the LCT design. Chapter 5.0 is the reference documents list. Chapter 6.0 is a configuration control provided for errata to be corrected in this document. The specification concludes in Appendix A (provided separately) with various test patterns, test messages and other data that can be used to verify the operation of selected functions within an LCT implementation.

EIA Standard RS-449, "General Purpose 37-Position and 9-Position Interface for Data Terminal Equipment and Data Circuit-Terminating Equipment Employing Serial Binary Data Interchange", dtd November 1977

EIA Standard RS-470, "Telephone Instruments With Loop Signaling for Voiceband Applications", dtd January 1981

Federal Communications Commission, Part 15, "Radio Frequency Devices", Title 47 CFR, dtd 1 October 1984

Federal Standard 1015, "Telecommunications: Analog to Digital Conversion of Voice By 2400 bit/second Linear Predictive Coding", dtd 28 July 1983

Future Secure Voice System FSVS-210, Rev E "Signaling Plan - Interoperable Modes"

FSVS-258, Rev A, "FD/CIK Bit Formats"

FSVS-370, Rev C, "Key Storage Device (KSD-64A)"

IEEE Standard 269-1983, "Standard Method for Measuring Transmission Performance of Telephone Sets", dtd December 1983

IEEE Standard 661-1979, "Method for Determining Objective Loudness Rating of Telephone Connections", dtd November 1979.

KAG-30A/TSEC, "Compromising Emanations Standard for Cryptographic Equipment", dtd 1968

MIL-E-16400G, "Electronic, Interior Communication and Navigation Equipment, Naval Ship and Shore: General Specification For", Addendum - 1, dtd 1 December 1976

SECTION 5
REFERENCED DOCUMENTS

The following documents in effect at the date of this Intero
perability Specification form a part of this specification to the extent specified in this document. In the event of a conflict, this document shall take precedence on issues related to FSVS interoperability.

ANSI S3.7-1973, "Method for Coupler Calibration of Earphones", dtd 1973

CCITT Recommendation V.26 bis, "2400/1200 bit/s Modem Standardized for Use in the General Switched Telephone Network", dtd 1980

CCITT Recommendation V.26 ter, "2400 bit/s Duplex Modem Using the Echo Cancellation Technique Standardized for Use on the General Switched Telephone Network and on Point-to-point 2-wire Leased Telephone-Type Circuits", dtd 1984

Defense Communication Agency Circular DCAC-370-V165-1, "AUTOVON Basic and Special Purpose Telephone Subscriber Equipment", dtd December 1965

Defense Communication Agency Circular DCAC-370-V175-6, "System Interface Criteria", dtd September 1978

EIA Bulletin No. 12, "Application Notes on Interconnection Between Interface Circuits Using EIA RS-449 and EIA RS-232-C", dtd November 1977

EIA Standard RS-423, "Electrical Characteristics of Unbalanced Voltage Digital Interface Circuits", dtd December 1978

SECTION 2
PERFORMANCE CHARACTERISTICS

The STU-III/LCT subscriber terminal performance characteristics are specified in this chapter, first in terms of the various mode requirements and then by the major functions of the terminal. The chapter is segmented into nine sections. Section 2.1 defines each of the STU-III modes and the mechanisms for changing modes. Section 2.2 presents a breakdown of the major functions of the STU-III. Section 2.3 specifies the STU-III audio interface requirements and recommendations. Section 2.4 specifies the voice processor requirements and Section 2.5 specifies the data function of the STU-III/LCT. Section 2.6 addresses all COMSEC subsystem functions and is provided as a classified attachment. Section 2.7 specifies the modem performance requirements. Section 2.8 specifies the Network Interface requirements. Section 2.9 specifies the user interface requirements.

2.1 MODES OF OPERATION (MER)

All LCTs shall provide Plain Old Telephone Service (POTS), support secure voice and data traffic and interact with the Key Management Center (Type I only). The primary STU-III/LCT secure traffic mode shall be full duplex. The STU-III/LCT shall also incorporate a half duplex mode as specified in the FSVS Signaling Plan, FSVS-210 to serve as a backup option when telephone lines are encountered that cannot support full duplex operation (e.g., excessive echo, or echo suppressors that cannot be disabled).

The Asynchronous Data, Reception of Secure Dialing, and Bit Error Rate Testing (BERT) modes are discretionary options for the STU-III/LCT design; all other modes (POTS, Secure Voice, Secure Data, and transmission of Secure Dialing) are Minimum Essential Requirements (MERs) for the STU-III terminal operation. All STU-III terminals, Type I and Type II, shall be

A Basic Line Interface Terminal (LIT)

For product code 26, the following alpha character codes apply.

A Universal Single Line (two or four wire) Terminal (SECTEL 1000 Based)
B Universal Single Line (two or four wire) 4.8 CELP Terminal (SECTEL 1500 Based)

For product code 27, the following alpha character code applies

A Basic STU-III/MPT (Mobile Portable Terminal)

For product code 28, the following alpha character code applies:

A Basic Multi Media Terminal (MMT) (secure voice at 2.4 kbps LPC, 4.8 kbps CELP, & 9.6 kbps MRELP and secure data with rates upto 9.6 kbps).

For product code 32, the following alpha character code applies.

A Basic Line Interface Terminal (LIT) Configuration

XXXXXXXX is an eight digit production number assigned sequentially by each manufacturer for each product code. Prototypes/FA will start with 00000001; Production units will start with 00001000 with no gaps or duplication.

The information that must be bar coded shall consist of a total of 14 characters and shall be in the following format:

"STUannXXXXXXXX" where the codes are defined as above. The bar code symbology shall be bar code 39 as per MIL-STD-1189.

interoperable with all other STU-III terminals in the interoperable signaling modes as specified in FSVS-210 when properly keyed.

2.1.1 Mode Descriptions

The following is a brief description of the LCT modes of operation:

2.1.1.1 Secure Voice (MER)

The STU-III/LCT shall provide a 2400 bps secure voice capability in both full and half duplex transmission modes as a MER. The voice digitizer shall be bit stream compatible with the DoD standard LPC-10 algorithm as specified in Section 2.4. The half duplex mode shall include a voice-operated switch (VOX) mechanism to control the half duplex transmission. The STU-III/LCT may also incorporate additional secure voice modes. If the STU-III incorporates additional secure voice modes, it shall diverge from the interoperable MER mode as defined in FSVS-210.

2.1.1.2 Secure Synchronous Data (MER)

The STU-III shall be capable of transmitting synchronous data securely at 2400 bps in both a full and half duplex transmission mode as a MER. See Section 2.5.1 for a description of the data mode. Other data modes are allowed as options. The STU-III shall support a full or half duplex data device while operating in the full duplex secure data mode. The STU-III shall also be capable of supporting two separate half duplex data transfers, concurrently, (one transmit, one receive) while in the full duplex secure data mode. All vendors' Type I or Type II STU-IIIs shall be interoperable with all other STU-III terminals in the interoperable synchronous data mode, specified in FSVS-210.

2.1.1.3 Secure Asynchronous Data

For product code 14, the following alpha character code applies.

A Basic Voice Encryptor (voice only up to 4.8 CELP)

For product codes 15 and 16, the following alpha character code applies.

A Universal Merlin Terminal

For product code 17, the following alpha character code applies.

A Universal 64 Kbps Terminal Model 1700 (AT&T Custom ISDN Standard)

For product codes 21 and 31, the following alpha character codes apply.

A Universal Single Line 4.8 CELP Terminal
B Universal Multiline 4.8 CELP Terminal

For product codes 22 and 23, the following alpha character codes apply.

A Basic Cellular Terminal Configuration (CONUS)
B Enhanced Cellular Terminal (CONUS)
C Enhanced Cellular Terminal (OCONUS)

For product code 24, the following alpha character code applies.

A Basic Automatic Remote Secure Telephone Unit (STU-III/R)

For product code 25, the following alpha character code applies.

It is recommended that the STU-III be capable of supporting a full duplex, asynchronous data mode (see Section 2.5.2). There are three recommended interoperable full duplex asynchronous data modes: 2400 bps full-rate, 1200 bps full-rate, and 1200 bps half-rate.

The interface between the STU-III and the DTE will be either a 2400 bps asynchronous interface or a 1200 bps asynchronous interface. The LCT/LCT interface will be the 2400 bps synchronous interface specified in Section 2.5.

Section 2.5.2 provides the requirements for the three interoperable asynchronous data modes. The asynchronous interface should support a 10-bit character format, including 1 start bit, 8 data bits, and 1 stop bit.

2.1.1.4 Plain Old Telephone Service (POTS) (MER)

The STU-III terminals shall provide Plain Old Telephone Service (POTS) as a Minimum Essential Requirement (MER). When in POTS mode, all calls are established as a plain analog connection through the network employing standard dialing and other user features.

2.1.1.5 Secure Dialing (MER)

The STU-III/LCT shall provide the capability of transferring dialing information securely to a far-end STU-III (or Line Interface Terminal (LIT)). The STU-III shall support secure dial transmit signaling as specified in FSVS-210 as a MER. The STU-III shall transmit the secure signaling so that an LIT can accept and process the received dial information to provide an interface with other FSVS network elements. The reception of secure dial information is not a STU-III/LCT MER.

Secure dialing shall only be initiated while the terminal is in the secure voice or secure data mode. The transfer of dialing information shall be sent in one

A CONUS Single Line Two Wire

B CONUS Multiline (5-Two Wire Lines) Basic Terminal

C N/A (Formerly CONUS Four Wire Basic Terminal)

D N/A (Formerly Foreign Two Wire Basic Terminal)

E N/A (Formerly Foreign Two Wire with Multiline)

F N/A (Formerly Foreign Four Wire Basic Terminal)

G Universal Single Line Two and/or Four Wire Basic Terminal*

H Universal Single Line Two Wire Basic Terminal

J Universal Single Line Four Wire Basic Terminal

K Universal Multiline (5-Two Wire Lines) Basic Terminal

L Universal Single Line (Two or Four Wire) Terminal (SECTEL 1500)

M Universal Multiline (5-Two Wire Lines) Terminal (SECTEL 1500)

N Universal Single Line 4.8 CELP Terminal

P Universal Multiline (5-Two Wire Lines) 4.8 CELP Terminal

*NOTE: Two and/or Four Wire: Both two and four wire operation is possible, however simultaneous two AND four wire operation is possible with GE/RCA (Dual Homed), whereas the user must set software straps for two OR four wire operation on the Motorola terminal. No external adapter is required.

For product codes 12 and 13, the following alpha character codes apply.

A Basic Line Encryptor (secure data only STU-III with rates up to 9.6 kbps; no voice)

B Enhanced Line Encryptor (secure data only STU-III with rates up to 14.4 kbps; no voice) direction; the reception of normal secure voice or secure data traffic in the other direction shall continue undisturbed except that LPC voice frames may be replaced with LPC silent frames for the duration of the secure dialing transfer.

2.1.1.6 Auto-Secure on Receive (MER)

The STU-III/LCT shall provide an "Auto-Secure on Receive" strap option to the users as a MER. This option shall enable the terminals to immediately and automatically initiate a secure call setup when the user goes off hook to answer an incoming call if the CIK is connected. Once this secure traffic is established, the terminals shall operate normally and permit a user abort (activating a Non-Secure control) or failed call or provide other options available to a normally strapped STU-III. If the CIK is not connected, the STU-III shall operate in the POTS mode only.

2.1.1.7 Plaintext Inhibit (MER)

The STU-III terminals shall provide a "Transmit Plaintext Inhibit" strap option as a MER. When this option is selected, the STU-III shall not enable any terminal microphone(s) until the terminal is in the secure mode. When placing or receiving calls, the terminal shall enable the handset receiver, but the microphone shall be muted. The STU-III shall initiate the secure call when the user activates a Secure control. During the secure call or call setup, the STU-III shall allow the call to be interrupted as in a normally configured STU-III; however, the microphone shall only be enabled during secure operation. If the Non-Secure control is activated to abort a secure call or in response to a failed call, the STU-III shall interrupt the call and shall only enable the handset receiver. The microphone shall remain muted. Transmit plaintext inhibit and autosecure on receive may be used in combination. If the CIK is not connected, the STU-III shall remain in the POTS mode with the microphone disabled.

14	AT&T TYPE II Voice Encryptor
15	AT&T Type I STU-III Merlin Terminal
16	AT&T Type II STU-III Merlin Terminal
17	AT&T Type I STU-III 64 Kbps Terminal
20	Motorola Type I LCT
21	Motorola Type II LCT
22	Motorola Type I Cellular
23	Motorola Type II Cellular
24	Motorola Type I Automatic Remote STU-III (STU-III/R)
25	Motorola Type I Line Interface Terminal (LIT)
26	Motorola Type I STU-III/A
27	Motorola Type I STU-III/MPT
28	Motorola Type I Multi Media Terminal
29	Motorola STU-II/B
30	GE/RCA Type I LCT
31	GE/RCA Type II LCT
32	GE/RCA Type I Line Interface Terminal (LIT)
40	Northern Telecom Type I LCT

NOTE: The GSN applies to the "basic unit" less any attachments which are optional. Therefore, the appropriate alpha code must be used to order the "basic unit". The user should specify "with Multiline" or use the National Stock Number (NSN) or vendor Model No. to order any attachment.

For product codes 10, 11, 20, 29, 30, and 40, the following alpha character codes apply.

2.1.1.8 Bit Error Rate Testing (BERT) (MER-Optional Mode)

The Bit Error Rate Test (BERT) is a recommended, discretionary option for the STU-III/LCT design. However, to ensure interoperability, a terminal providing a BERT mode shall provide this mode as specified in FSVS-210. If the LCT provides a BERT capability, it shall support both full and half duplex transmission mode operation. The BERT mode shall be entered from either the speech mode or from the data mode, and use of this mode shall enable the STU-III to test the transmission performance of the STU-III/ STU-III link. In full duplex, both STU-IIIs shall transmit encrypted zeros for the duration of the mode. In the half duplex BERT mode, the initiating terminal shall transmit a nominal ten second burst of encrypted zeros; the far-end terminal will follow this with a ten second burst of encrypted zeros.

2.1.1.9 KMC (Type I-MER)

The Type I STU-III shall be capable of interacting with the KMC in order to receive rekeying messages and/or the Compromised Key List (CKL). The STU-III/ KMC signaling shall be as specified in FSVS-210. At the conclusion of a successful transfer, the STU-III shall update the previously-stored FIREFLY II keying material and/or the previously-stored CKL using the material received from the KMC as specified in Section 2.6.5.1.

2.1.2 Mode Changing and Control (MER)

The primary mode of the STU-III/LCT is 2400 bps, full duplex secure voice. In general, if conflicts occur in attempting to setup or change modes, the LCT shall default to the primary mode. The STU-III/LCT design shall incorporate the capability of changing modes as specified in this section.

2.1.2.1 Full/Half Duplex Selection (MER)

B. Nomenclature Markings

The STU-III/LCT nomenclature shall be as follows:

Type I must display - "Controlled Cryptographic Item" and "EC"

Type II must display - "Endorsed-for-Unclassified Cryptographic Item" and "Subject to Export Control"

Both must display - "GSN:", then 14 characters in the following format:

GSN: STUannXXXXXXXX

28

Where:

GSN indicates that this is the Government assigned Government Serial Number.

STU indicates that this is a STU-III equipment.

a is an alpha character code which provides terminal configuration information tied to unique "nn" product codes as shown below. The description may vary from one product code to another.

nn is a two digit numeric product codes which indicates the vendor, Type, and STU Family of Equipment (LCT, LIT, Cellular, STU-III/R, STU-III/MPT, STU-III/A, STU-III). Those currently assigned by the Government are:

10 AT&T Type I LCT

11 AT&T Type II LCT

12 AT&T Type I Line Encryptor

13 AT&T Type II Line Encryptor

The LCT shall incorporate the capability to support full and half duplex secure operations. The primary mode will be full duplex. As such, the LCT shall require that the user manually selects half duplex when desired. The selection shall be made while the LCT is either on-hook or is in the POTS mode of operation (i.e., during an initial clear call, or following the activation of a clear call control such as a Failed Call or User Abort). The STU-III shall incorporate the initial full or half duplex signaling as specified in FSVS-210. Consistent with that, only one of the terminals has to select half duplex in order for the call to proceed through half duplex call set up into a secure half duplex call.

2.1.2.2 Initial Modem Training Mode Selection

The FSVS Signaling Plan, FSVS-210 identifies a capability to permit the coordinated selection of a special mode (e.g., 4800 bps) prior to normal 2400 bps modem training. This is not a MER for the STU-III/LCT; it is a discretionary option that the LCT may incorporate. This option allows two LCTs to determine if there is a common preferred mode. If the common mode exists, the LCTs will automatically change to that mode.

2.1.2.3 Secure Mode Selection (MER)

As specified in FSVS-210, the 2400 bps, interoperable, MER mode requires that the STU-III/LCT initially enter either the secure voice or secure data mode. The LCT shall be capable of changing from one secure mode to another without the user aborting first to a clear, plaintext analog call. These changes shall include:

- Entering from the secure voice or data to secure dialing mode and back to secure voice or data (only affects the transmission in one direction).

- Entering from the secure voice or data to Bit Error Rate Test (BERT) mode and back to secure voice or data (BERT is a discretionary option for the LCT).

SECTION 3
DESIGN CHARACTERISTICS

This section provides the physical, environmental, safety and electromagnetic requirements and recommendations for the STU-III.

3.1 PHYSICAL CHARACTERISTICS

3.1.1 Physical Requirements

3.1.1.1 STU-III/Deskset (MER)

The size, weight, power consumption, form factor, and environmental design of the STU-III or the deskset, if a separate item, shall be suitable for use at a desk or in other office configurations. The terminal shall have an attractive appearance. Acoustic and thermal emanations shall be limited to those appropriate for a quiet, office environment, and acoustic noise shall not exceed the requirements of MIL-STD-1472, Curve NC-35.

3.1.1.2 Nomenclature (MER)

A. Physical Location

The nomenclature information must be displayed on a Type I and Type II terminal, somewhere on the exterior of the unit (bottom, front, etc.), with the exact location at the option of the manufacturer. The method of marking the nomenclature shall be permanent. At the option of the manufacturer, the nomenclature may be on a nameplate (adhesive-backed or attached via screws) or may be stamped or engraved on a surface.

- Changing from secure voice to secure data, or from secure data to secure voice.

- Aborting from any secure mode to an analog clear call (after which other modes may be selected, as appropriate).

With the exception of the User Abort (or Failed Call) sequence, each of these changes requires a crypto resynchronization. Indication of the mode change will be identified in the Message ID (MID) field of the cryptosync message. If an incompatible mode is selected, the LCTs at both ends will default to the 2400 bps MER secure voice mode.

2.2 TERMINAL FUNCTIONAL OVERVIEW

Figure 2-1 is a functional block diagram of the STU-III terminal. The intent of the diagram is to present the breakdown of functional specifications for the STU-III terminal. It is not intended to reflect any specific equipment design. The remainder of this chapter provides a specification for each of the major functions shown in the figure.

2.3 AUDIO INTERFACE

Section 2.3.1 identifies specific secure mode MERs for the audio interface. Sections 2.3.2 and 2.3.3 provide secure mode and POTS mode recommendations, respectively, for a more comprehensive audio interface. Additional requirements for the audio interface performance are inherent in the end-to-end secure speech intelligibility and quality requirements of Section 2.4.

2.3.1 Secure Mode Requirements (MER)

The audio interface requirements for the secure voice operation are provided below.

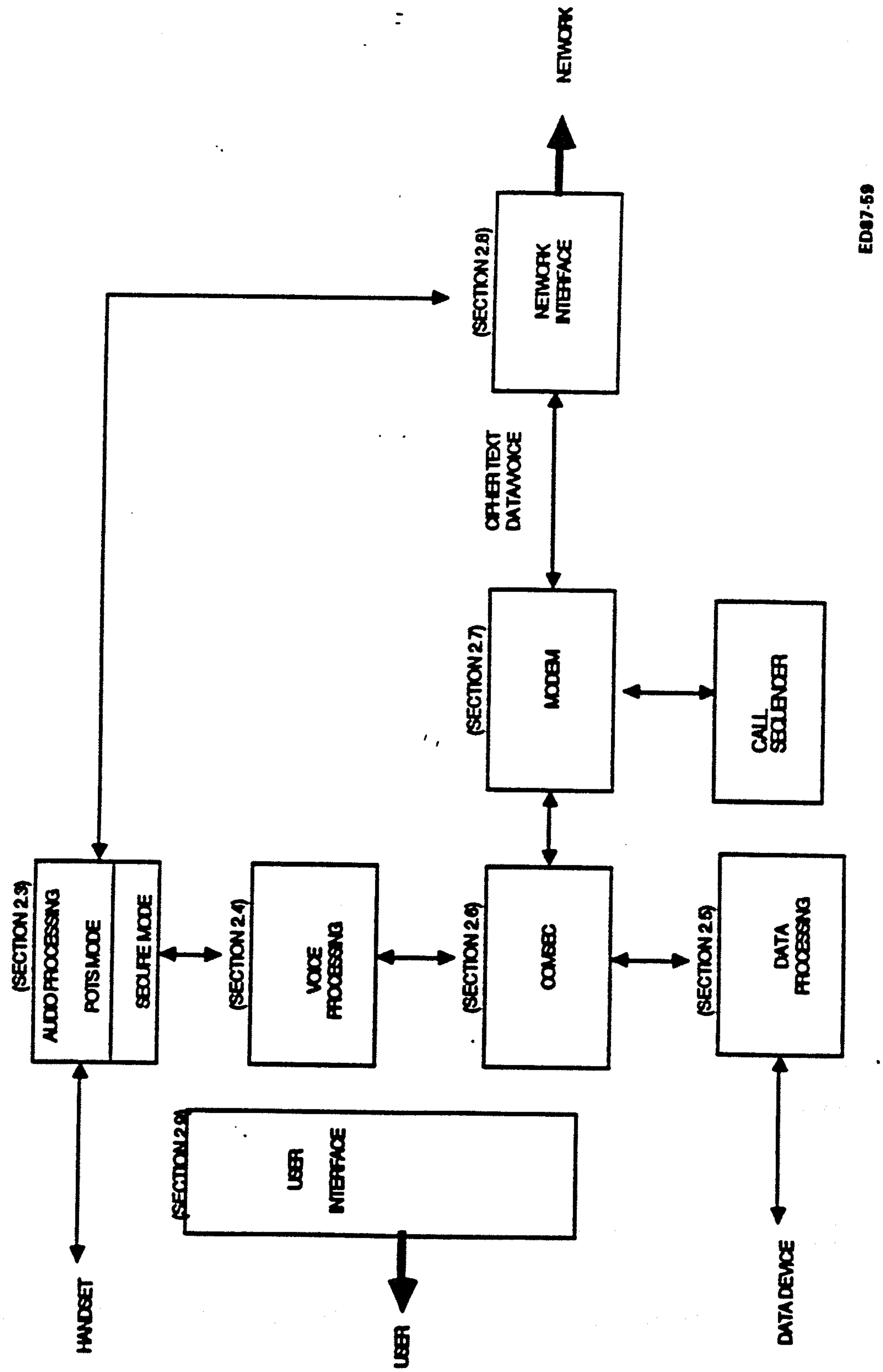

Figure 2-1. Terminal Functional Block Diagram

2.10.4.9. As an option, the terminal may warn the user if a Maximum Security Level is entered which is less than the corresponding Minimum Security Level for the same keyset, since such a combination will cause all secure call setup attempts to fail.

2.10.5 Users Manual

A list of Government Security Guidelines (supplied by the Program Office) must be included in the Users' Manual. This list provides users with areas of concern on the installation, programming and operation of the access control features that must be considered to maintain security.

A. Transmit Gain Level (MER):

1) Requirement

Using a 600 Hz sinusoidal input, a sound pressure level (SPL) of 111 dB ± 3dB into the microphone shall fully load the RMS value in the speech algorithm.

2) Measurement Procedure

(a) The artificial mouth (B&K Instruments Inc., type-4219 artificial voice, or equivalent) shall meet the requirements of Section 5.2 of IEEE Standard 269-1983. The artificial mouth output shall be measured at 25 ± 1mm from the mouth reference plane (lip ring). The artificial mouth output shall be adjusted to a sound pressure of 111 ± 1dB peak relative to 20 μPa at 600-Hz. The 0o-incidence, free-field response of the microphone shall be used in determining the sound pressure.

(b) The telephone handset shall be mounted in a modal position (per Figure 1 of IEEE Standard 269-1983) on a test head designed for telephone measurements (B&K Instruments, Inc.,type-4905 telephone test head, or equivalent). The receiver shall be acoustically terminated on the artificial ear. With the handset in the modal position, the location of the microphone relative to the lip ring of the artificial mouth is determined by the shape of the handset.

B. Receive Gain Level (MER):

1) Requirement

A fully loaded RMS value received as input to the speech synthesizer shall produce a SPL of 99 dB ± 3dB in the earpiece.

2.10.4.3. The initial setting of a Maximum Security Level and any subsequent modification shall be allowed only when the associated keyset has been activated by its Master CIK.

2.10.4.4. A Maximum Security Level shall apply only to the keyset which was activated by its Master CIK when the Maximum Security Level was entered, added or otherwise changed or modified.

2.10.4.5. As an option, terminals which provide a Maximum Security Level for a keyset may also allow the Maximum Security Level to be enabled and disabled, but such enabling and disabling shall be allowed only when the associated keyset has been activated by its Master CIK.

2.10.4.6. When the capability is provided to enable and disable the Maximum Security Level, such enabling and disabling shall be performed without affecting the enabled or disabled state of the ACL or Minimum Security Level.

2.10.4.7. Unless the Maximum Security Level is disabled for the active keyset, secure call setup shall be completed only when the security level of the remote STU-III is less than or equal to the Maximum Security Level specified. It should be noted that the comparison is against the far end terminal's security level, not the common security level established for the call.

2.10.4.8. The Maximum Security Level shall allow settings for the following levels:

1. Unclassified
2. Confidential
3. Secret
4. Top Secret

2) Measurement Procedure

(a) The artificial ear shall be the IEC coupler for supra-aural earphones per ANSI S3.7-1973. The pressure response of the microphone shall be used in determining the sound pressure generated in the coupler by the receiver.

(b) LPC parameters, representing a voiced waveform with a fuly loaded RMS value, shall be applied to the LPC-10 synthesizer. The synthesized digital bit-stream, representing the fully loaded RMS input shall be applied to the input to the D/A converter. For STU-III's with a volume control, the SPL measurement shall be made at the recommended nominal setting of the volume control.

C. <u>Effect of Microphone (MER)</u>: The handset microphone, while in use, shall not degrade the frequency response between the handset and the A/D converter by more than ± 3 dB between 300 Hz and 3600 Hz, and, +3 dB or -5 dB at 100 Hz. Microphone puff and breath noise suppression shall be provided by the equivalent of a wire mesh screen a minimum of 1/4" from the diaphragm having a mesh of 25 wires per inch or greater.

D. <u>Effect of Earpiece (MER)</u> The handset earpiece while in use shall not degrade the frequency response between the handset to A/D converter to D/A converter to handset output by more the ± 3 dB between 200 and 3000 Hz, nor more than ± 6 dB between 3000 and 3600 Hz.

E. <u>Frequency Response (MER)</u>:

(1) The handset to A/D converter inputs (without microphone) relative to the average level in the vicinity of 1004 Hz shall exhibit the following characteristics:

At or Below 50 Hz - Down 16 dB minimum

At 100 Hz - Down 15 dB maximum

2.10.3.5. A Minimum Security Level shall be capable of being enabled and disabled, but such enabling and disabling shall be allowed only when the associated keyset has been activated by its Master CIK.

2.10.3.6. A Minimum Security Level shall be capable of being enabled and disabled without affecting the enabled or disabled state of the ACL or Maximum Security Level.

2.10.3.7. When the Minimum Security Level is enabled for the active keyset, secure call setup shall be completed only when the common security level established for the secure call meets or exceeds the Minimum Security Level specified.

2.10.3.8. The Minimum Security Level shall allow settings for the following levels:

1. Unclassified
2. Confidential
3. Secret
4. Top Secret

2.10.3.9. As an option, the terminal may warn the user if a Minimum Security Level is entered which is greater than the corresponding Maximum Security Level for the same keyset, since such a combination will cause all secure call setup attempts to fail.

2.10.4 Maximum Security Level (Option)

2.10.4.1. A Maximum Security Level may be provided as an option for each keyset in a terminal independently of the provision of the ACL and Minimum Security Level defined above.

2.10.4.2. A Maximum Security Level shall apply to a single keyset.

At 200 Hz - Down 2 dB maximum

In Band (300 Hz to 3000 Hz) - ±.75 dB ripple maximum

At 3200 Hz - Down 1.5 dB maximum

At 3600 Hz - Down 6 dB maximum

At 4000 Hz - Down 12 dB minimum

Above 4600 Hz - Down 32 dB minimum

Above 8000 Hz - Down 35 dB minimum

This frequency response correlates to the frequency response of a CODEC chip. A filter with sharper cutoffs may be used.

(2) The handset to A/D converter to D/A converter to Handset inputs (without microphone or earpiece but including (sin x)/x compensation) relative to the average level in the vicinity of 1004 Hz shall be:

At 100 Hz - Down 15 dB maximum

In Band - 1 dB ripple

At 3600 Hz - Down 12 dB maximum

(3) From any handset to any other handset path (without microphone) from 100 Hz to 3600 Hz shall be +0.5, -1.0 dB, referenced at 1004 Hz, with smooth roll-off beyond 3600 Hz.

2.3.2 Secure Mode Recommendations

The performance parameters listed below, measured at cable lengths of ten feet, are recommended for the handset/converter interface, for either a conventional handset (microphone and earpiece) or a handheld microphone with an external speaker, to ensure good audio performance.

2.10.2.10. When going secure, a terminal shall compare the KMID of the far end terminal to the CKL. The CKL shall be checked before the ACL and Security Levels defined below.

2.10.2.11. If transfer of an ACL to and from a KSD is allowed, the format shall be as specified in FSVS-259.

2.10.2.12. Wildcard characters in the DAO codes and KMIDs on an ACL are prohibited.

2.10.2.13. Duplicate entries of DAO codes or KMIDs tagged to the same keyset shall not be allowed on an ACL.

2.10.2.14. An ACL shall be covered with a cryptographic checksum. The checksum shall be generated over the entire list, to include any header and tag bits, using the KG. The checksum shall be exactly 8 bytes long.

2.10.3 Minimum Security Level (Option)

2.10.3.1. A Minimum Security Level may be provided as an option for each keyset in a terminal independently of the provision of the ACL and Maximum Security Level defined below.

2.10.3.2. A Minimum Security Level shall apply to a single keyset.

2.10.3.3. The initial setting of a Minimum Security Level and any subsequent modification shall be allowed only when the associated keyset has been activated by its Master CIK.

2.10.3.4. A Minimum Security Level shall apply only to the keyset which was activated by its Master CIK when the Minimum Security Level was entered, added or otherwise changed or modified.

A. Sidetone Level: The sidetone SPL at the earpiece should be 6 dB ± 4dB SPL below the transmit speech level into the microphone. The sidetone level should be measured according to IEEE 269-1983, using the test circuit of Figure 3D of that standard.

(1) For identical A/D and D/A converter loading, the bridging level among handsets should be able to be set to 0 dB +0/-4 dB relative to the same level received from a remote terminal in the secure mode.

(2) The sidetone level should be set to 0 dB for the conditions specified in (1). The sidetone for each handset should be capable of being independently disabled.

B. Crosstalk: The audio crosstalk of the receive path (D/A converter to handset) into the transmit path (handset to A/D converter) should be at least 45 dB down at 1004 Hz with the microphone replaced by a load with equivalent impedance. With the microphone installed, and the receiver acoustically terminated, the crosstalk should be down 43 dB minimum. Levels should be measured at the A/D converter input relative to the level at the D/A converter output at a receive signal level that produces a SPL of 90 dB in the IEC coupler of the artificial ear when measured as specified in EIA RS-470, Section 4.1.2.2.(3).

C. Signal to Noise Ratio (SNR): The SNR should be a minimum of 60 dB between any acoustically terminated handset and the A/D converter input at full A/D converter loading using 1004 Hz test tone and noise band of 100 Hz to 3600 Hz. The SNR should be a minimum of 55 dB between any handset to the A/D converter, a digital loopback to the D/A converter and back to the Handset at full A/D - D/A converter load.

D. Distortion:

2.10.2 Access Control List (Option)

2.10.2.1. As an option, a STU-III may provide an Access Control List for a keyset in accordance with the requirements of this paragraph.

2.10.2.2. An Access Control List shall be associated with a single keyset.

2.10.2.3. Initial entry of data, additions, deletions, changes and other modifications to the ACL shall be allowed only when the associated keyset has been activated by its Master CIK.

2.10.2.4. An ACL shall support DAO codes and KMIDs.

2.10.2.5. An ALC shall provide storage for 500 entries, where an entry may be either a DAO code or a KMID.

2.10.2.6. DAO codes and KMIDs on an ACL shall apply only to the keyset which was activated by its associated Master CIK when the DAO codes and KMIDs were entered, added or otherwise changed or modified.

2.10.2.7. An Access Control List shall be capable of being enabled and disabled, but such enabling and disabling shall be allowed only when the associated keyset has been activated by its Master CIK.

2.10.2.8. An ACL shall be capable of being enabled and disabled without affecting the enabled or disabled state of the Minimum Security Level or Maximum Security Level specified below.

2.10.2.9. When the Access Control List associated with the activated keyset is enabled, secure call setup shall be completed only with remote terminals whose RTC contains a KMID or DAO code found on the ACL.

(1) The total distortion should be at least 40 dB below a 300 Hz signal level between any handset and the A/D converter input at 110% full A/D converter loading (without microphone). Increasing the input signal up to the power supply levels should result only in a flat-topping of the signal at the A/D converter input.

(2) The total distortion should be at least 40 dB below a 1004 Hz signal level between any handset to the A/D converter, a digital loopback to the D/A converter, and back to the handset at full A/D converter loading (without microphone).

(3) The total distortion between any handset and the A/D converter when the microphone is driven with a 1004 Hz signal should be at least 34 dB (2 percent) below the source when the microphone is driven by a 100 dB Sound Pressure Level (SPL), and at least 26 dB (5 percent) below the source when driven by a 116 dB SPL.

E. Gain/Level: As a recommendation, the gain/level requirements of Section 2.3.1.B.1 for the earpiece, should be met with the level being adjustable over a range of +12 dB ± 1 dB to -12 dB ± 1 dB range (with a 2 dB maximum step size) using an earpiece that satisfies the requirements of ON270702. An M87 earpiece may be used.

F. Isolation: When on-hook, each handset microphone (replaced by a load with equivalent impedance), earpiece, and hook switch should be isolated by at least 100 dB at 1004 Hz from any other handset and the A/D converter.

G. Loopback: When a loopback capability is implemented at the handset/converter interface, the following recommendations should be satisfied.

elements: an Access Control List (ACL), a Minimum Security Level, and a Maximum Security Level, each of which may be offered independently. When an access control system is provided, it shall be in accordance with the requirements specified below.

2.10.1 Access Control for Terminals with Multiple Independent Keysets.

2.10.1.1. When provided, an ACL, a Minimum Security Level and a Maximum Security Level shall be associated with a single keyset.

2.10.1.2. Terminals with multiple independent keysets, provided in accordance with FSVS-220, Section 2.6.9.2, may provide an ACL, a Minimum Security Level and a Maximum Security Level for each keyset.

2.10.1.3. Providing an ACL, a Minimum Security Level, or a Maximum Security Level for one keyset in a terminal shall not imply a requirement to provide the same access control features for other keysets in the terminal. The choices in providing an ACL, a Minimum Security Level and a Maximum Security Level are options that may be exercised independently for each keyset.

2.10.1.4. When an ACL, a Minimum Security Level or a Maximum Security Level are provided for a keyset, they shall operate exclusively in conjunction with that keyset, regulating the access to secure call setup with that keyset and shall have no effect on secure call setup with other keysets in the terminal.

2.10.1.5. The ACL, Minimum Security Level and Maximum Security Level for a keyset and the processing performed to enter, alter, transfer or otherwise use the access control features provided shall be independent of the processing performed for similar features associated with other keysets within the terminal.

2.10.1.6. All ACLs in the STU-III shall be zeroized when the terminal is zeroized (either by activation of the zeroize switch or anti-tamper switches).

(1) The D/A analog output should be capable of being looped back to the A/D input. The gain of this path should be 0 dB ±3 dB at 1004 Hz. When disabled, the gain of this path should drop by 60 dB minimum.

(2) The A/D output should be capable of being digitally looped back to the D/A input. The gain of this loopback path should be 0 dB at 1004 Hz.

2.3.3 POTS Mode Recommendations

The following recommendations for the audio interface between the handset and the telephone line are designed to ensure good quality audio performance.

A. Transmit Response: The transmit response of a telephone is a measure of its acoustic-to-electrical transfer characteristics. To define the transmit response, the output voltage or loudness level, the frequency response, and the regulation over a given set of loop conditions are specified.

1. Measurement Procedure: The transmit characteristics should be measured according to IEEE Standard 269-1983, using the test circuit of Figure 3B of that standard. The conditions listed in EIA RS-470, Section 4.1.1.2 apply to the measurement procedure.

2. Transmit Objective Loudness Rating (TOLR): The TOLR should be determined for each of the loop conditions given in EIA RS-470 Section 4.1.1.2.(2). Any of the methods described in IEEE Standard 661-1979 may be used.

   The loudness rating should be determined over the frequency range 300 to 3300 Hz, as specified in IEEE Standard 661-1979. The TOLR should fall between the upper and lower limits given in EIA RS-470 Table 1. It is desirable that the TOLR have the mean upper and lower limits given in EIA RS-470 Table 2.

Table 2-10. ID Display Messages

ID Field	Display Format*
Class	Decimal Equivalent (e.g., "Class = 4")
Classification	Standard Display per Table 2-9 (e.g., "Classification = TS")
Key ID Number	Decimal Equivalent of Binary Number (e.g., "Key ID Number = 120)
Key Mat. Exp. Date (Type I calls only)	Decimal Equivalent of Hex Digits (e.g., "Exp. Date = 05/88").
Authentication Data	ASCII Characters

* The display format of each field is left to the discretion of the vendor. (Recommended display formats are provided in parentheses).

3. <u>Transmit Frequency Response</u>: The frequency response in the POTS mode at the handset/telephone line inter-face relative to the average level in the vicinity of 1004 Hz (pass band region) should be as follows:

At 50 Hz - Down 20 dB minimum

At 200 Hz - Down .5 dB maximum

In Band (300 Hz to 3000 Hz) - ±.5 dB ripple

At 3200 Hz - Down .5 dB maximum

At 3600 Hz - Down 4 dB maximum

Above 4000 Hz - Down 16 dB minimum

Above 5000 Hz - Down 45 dB minimum

This frequency response correlates to the frequency response of a CODEC chip. A filter with sharper cutoffs may be used.

B. <u>Receive Response</u>: The receive response of a telephone is a measure of its electrical-to-acoustic transfer characteristics. To define the receive response, the acoustic output or loudness level, the frequency response, the regulation over a given set of loop conditions, and the distortion are specified.

1. <u>Measurement Procedure</u>: The receive characteristics should be measured according to IEEE Standard 269-1983, using the test circuit of Figure 3C of that standard. The conditions (1) - (6) listed in EIA RS-470, Section 4.1.2.2 apply to the measurement procedure.

2. <u>Receive Objective Loudness Rating (ROLR)</u>: The ROLR should be determined for each of the loop conditions given in EIA RS-470 Section 4.1.2.2(2). Any of the methods described in IEEE 661-1979 may be used. The

2.9.4 POTS Mode Display

2.9.4.1 Type I Display Requirements

A. When off-hook in the POTS mode, the Type I terminal shall display "NONSECURE" on the first line of the display.

B. When the CIM has been decoded (see section 2.6.4.4), those STU-IIIs being prompted shall display the appended message as indicated in Table 2-7, when the CIK is inserted. The CIK must be inserted for the user to clear the prompt message from the display. This requirement applies to Type I terminals only. Type II terminals do not receive a CIM.

2.9.4.2 Type II Display Recommendation

It is recommended that the Type II terminal display "UNPROTECTED" on the first line of the display when off-hook in the POTS mode.

2.9.5 Auto-Answer in the Data Mode

When the STU-III supports an auto-answer by a DTE in the data mode, (i.e., the STU-III answering a call before the user takes the handset off-hook) the STU-III shall provide the far-end authentication data to the near-end DTE.

2.10 ACCESS CONTROL SYSTEM

As an option, the STU-III may offer an access control system in association with a keyset to limit secure call setup to those remote STU-IIIs which meet the access control requirements. An access control system contains three loudness rating should be determined over the frequency range 300 to 3300 Hz, as specified in IEEE 661-1979. The ROLR should fall between the upper and lower limits given in EIA RS-470 Table 3. It is desirable that the ROLR have the mean, upper and lower limits given in EIA RS-470 Table 4.

3. Receive Frequency Response: The receive frequency response for the 0-kft loop condition, recorded as specified in EIA RS-470, Section 4.1.2.2(5), should fall within the upper and lower limits of the curve shown in EIA RS-470, Figure 2.

C. Receive Distortion: The receive harmonic distortion should be less than five percent. The receive harmonic distortion should be determined by applying a 1004-Hz signal from the generator to the telephone while it is connected as described in EIA RS-470 for the 0-kft loop condition. A harmonic distortion measuring set should be connected to the output of the microphone amplifier to read the percent distortion.

D. Isolation: The Plaintext interface should provide the electrical isolation of 100 dB minimum at 1004 Hz between any handset (with the microphone replaced by a load with equivalent impedance) and the telephone line when the plaintext mode is not specifically enabled.

E. Additional POTS Mode Recommendations: It is recommended that the STU-III satisfy the additional EIA RS-470 requirements regarding:

1) Telephone Noise
2) Loop Supervision
3) Address Signaling
4) DC and AC Current Characteristics
5) Alerting Characteristics
6) Maintenance Related Characteristics.

If the far-end terminal class is 6, the characters in the special ID field of the received authentication data are displayed using the criteria specified in Section 2.9.2.2. The STU-III shall continuously display the next subfield (depending on the length of the second display line) of authentication data of the far-end terminal that follow the special ID field, on the second (and any subsequent) line of the display during the secure call.

If the far-end terminal class is not 6, the STU-III shall continuously display the first subfield of authentication data of the far-end terminal, that follow the eight-character DAO code field, on the second line of the display during the secure call. If the STU-III has more than two display lines, the STU-III may continuously display additional characters of authentication data of the far-end terminal on the third and subsequent display lines during the secure call.

The far end authentication data (excluding the special ID field for Class 6 terminals, which may be optionally displayed if the display criteria of Section 2.9.2.2 are met) shall be available for line by line rescrolling on the second (and any subsequent) line of the terminal display at the request of the local user during a secure call.

2.9.3.2 Far-End ID Data

The entire ID data (excluding the special ID field for far-end Class 6 terminals, which may be optionally displayed if the display criteria of Section 2.9.2.2 are met) for the far end terminal shall be available for line by line rescrolling on the second (and any subsequent) line of the terminal display at the request of the user during a secure call.

2.9.3.3 Near-End ID Data

The STU-III shall scroll the entire near-end user ID data on a line-by-line basis when the user activates a control requesting this feature (only while the STU-III is in POTS mode and if the CIK is inserted).

2.3.4 Push-to-Talk Handset Requirements (MER)

As an ancillary device, a handset shall be offered which provides a "push-to-talk" (PTT) capability. This handset shall, as a minimum, contain a momentary hand or finger-operated switch which, in the normal or unactivated state, electrically opens the microphone circuit in the handset in all clear and secure voice and data modes of operation. When activated, this switch shall allow for normal clear and secure voice operations.

With the PTT activated, the STU-III shall continue to meet all of the audio interface performance requirements specified in paragraph 2.3 of FSVS-220.

It is not intended that the PTT handset control any terminal functions other than disabling of the microphone circuit. For example, control of Half-Duplex Transmitter/Receiver functions by the PTT functions or push-to-listen capabilities are neither required nor desired.

2.3.5 Push-to-Talk Handset Recommendations

It is recommended that the Push-to-Talk Handset provide the following features:

a. A push-to-talk switch which provides additional on-hook audio protection by shorting the microphone as well as opening the circuit to the telephone when the PTT is in the unactived condition.

b. A push-to-talk button physically located near the receiver element which could be operated by the thumb.

2.4 VOICE PROCESSOR

D. Display Location: When the special access information is displayed, it shall be to the right of the classification, on the top line of the display.

E. Examples:

Terminal A Special ID field characters	Terminal B Special ID field characters	Common Display on Terminals A and B
!ABC!DEF (end)	!ABC!DEF (end)	ABC DEF
!AB!DEF! (end)	!AB!DE! (sp) (end)	AB
!ABC!XYZ (end)	!XYZ!ABC (end)	ABC XYZ (or XYZ ABC)
!ABC!DEF (end)	!XYZ! (sp) (sp) (sp) (end)	(no display)
!A!BC!DE (end)	!A!DE! (sp) (sp) (end)	A DE
ABC!DEF (sp) (end)	!ABC!DEF (end)	ABC!DEF on Terminal B (no display on A)
not class 6	ABCXYZ (sp) (sp) (end)	ABCXYZ on Terminal A (no display on B)

In the above examples, "(sp)" is defined as an ASCII "space" character, or Hex 20. The 8-character class 6 field will always be followed by a 9th character, the ASCII "end" or Hex 00, defining the end of the field. On the display, a space is used to separate special accesses from each other and from the classification display.

2.9.3 Terminal ID and Authentication Data Display (MER)

2.9.3.1 Far-End Authentication Data

The STU-III shall automatically scroll the authentication data of the far-end terminal (excluding the DAO code and special field) one time at a minimum, on the display during call set-up.

The STU-III shall incorporate a 2400 bps voice processor that can operate in the full duplex and half duplex modes over the communication channels that would be encountered by calls within CONUS. The voice processor shall be bit-stream compatible with the government standard LPC-10(e) and must meet or exceed the MERs for intelligibility and quality as specified in Section 2.4.1.

The Government recommendation for the voice processor is LPC-10(e). The algorithm description of LPC-10 is provided in Section 2.4.3. The analysis and synthesis improvements to LPC-10, incorporated in LPC-10(e), are described in Section 2.4.4. A FORTRAN listing of LPC-10(e) is provided as an appendix to this document.

If voice processors alternative to the 2400 bps LPC-10(e) voice processor are implemented, they shall interoperate with the government LPC-10(e) with intelligibility and quality scores that meet or exceed those for the LPC-10(e). The bit stream for LPC-10(e) and alternative voice processors is given in Section 2.4.2.

2.4.1 LPC-10(e) Intelligibility and Quality Requirements

Paragraphs 2.4.1.1, 2.4.1.2, and 2.4.1.3 provide STU-III speech processor requirements for intelligibility and quality, and methods for measurement of these requirements. These requirements apply to all versions and vendors (i.e., each vendor's Type I or Type II terminal must meet the intelligibility and quality requirements when tested with his own or another vendor's Type I or Type II terminal).

2.4.1.1 Intelligibility (MER)

The STU-III intelligibility shall meet or exceed that of the Government-supplied LPC-10(e) algorithm when tested with standard Diagnostic Rhyme Tests (DRT) using the three male and three female speakers identified in Section 2.4.1.3 A. Test and evaluation procedures are defined in Section 2.4.1.3.

2.9.2.2 Special ID Field Display (Class 6)

When the far-end terminal class is 6, the STU-III shall use the first character in the far end special ID field and the first character in its own (local) special ID field (if one exists) to determine the display criteria for this field. The display criteria shall be as specified in the following sections.

A. Far-End Terminal Class 6 With !; Local Without: If the first character of the far-end special ID field is an exclamation point, "!", and if the local terminal does not have class 6 key or if the local terminal has class 6 key but does not have a "!" in the first character position of its own special ID field, then the local terminal shall not display any of the information in the remote terminal's special ID field.

B. Far-End Terminal and Local Terminal Class 6 With !: If the first character of the far-end special ID field is an exclamation point, "!", and if the local terminal has class 6 key with a "!" in the first character position of its special ID field, then it shall compare all of the remaining positions of the remote terminal's special ID field with its own and shall display only those special accesses which are common to both terminals. Special accesses will always be separated by exclamation points "!" or by the boundaries of the special ID field (end character, or 00 hex). Special accesses can be of any length up to 7 characters. Special accesses can be in any order. Matching special accesses shall be displayed continuously on the top line of the display to the right of the classification during the secure call.

C. Far-End Terminal is Class 6 Without !: If the far-end terminal is Class 6 and the first character in the far-end terminal's special ID field is not an exclamation point, "!", then all of the characters in the special ID field of the received authentication data shall be displayed continuously on the top line of the local display to the right of the classification during the secure call.

2.4.1.2 Voice Quality (MER)

The STU-III voice quality shall meet or exceed that of the Government-supplied LPC-10(e) algorithm when tested with standard Diagnostic Acceptability Measure (DAM) tests using the three male and female speakers identified in Section 2.4.1.3 A. Test and evaluation procedures are defined in Section 2.4.1.3.

2.4.1.3 Test Conditions (MER)

A. Speakers: The speakers to be used for the DRT and DAM tests shall be the following two sets:

RH/JE/CH - male
VW/KS/MP - female.

The standard tape of these speakers will be made available during the LCT test phase.

B. Test Method: The test method shall be as follows: A standard tape of these speakers will be provided by the Government for intelligibility and voice quality testing. This tape shall be played into the STU-III and the output recorded via test procedures reviewed and approved by the Government. In order to eliminate variability in scoring, all vendor's output tapes and the Government output tapes will be scored concurrently by the same laboratory.

C. Required Scores:

1) Tests between STU-IIIs built by same vendor: For both intelligibility and voice quality scoring, a passing score shall be achieved if the vendor's average score, plus any (listener) variance identified by the laboratory, equals or exceeds the Government's average score.

Table 2-9. Standard Security Classification Displays

Security Classification	Truncated Display	Full Display
Type I:		
Unclassified	UNCLASS	UNCLASSIFIED
Confidential	CONF	CONFIDENTIAL
Secret	SECRET	SECRET
Top Secret	TS	TOP SECRET
Type II:		
Unclassified	PROTECT	PROTECTED

2) Test between different vendors' STU-IIIs: For both intelligibility and voice quality scoring, a passing score will be achieved if the vendor's average score, plus any (listener) variance identified by the laboratory in scoring the vendor, equals or exceeds the Government's average score minus the (listener) variance on the Government score.

2.4.2 LPC-10 Bit Stream (MER)

The 2400 bps STU-III voice processing algorithm shall be bit stream compatible with LPC-10 (e) (the bit stream is defined in FED-STD-1015). Voiced frames shall be characterized by the first through tenth reflection coefficients, coded to 5,5,5,5,4, 4,4,4,3, and 2 bits, respectively. Amplitude and pitch information shall also be coded and a sync bit appended to each frame. Unvoiced frames shall be characterized by four reflection coefficients, amplitude and pitch parameters and a sync bit. The 21 available bits, allocated for the fifth through tenth reflector coefficients, shall be utilized to protect the amplitude information and the first four reflection coefficients.

Table 2-1 defines the allocation of the LPC-10(e) parameters in the 54-bit transmission frame. As a standard convention, the STU-III shall set the alternating 1/0 sync bit of the first speech frame of each secure voice traffic transmission to a "0".

2.4.3 LPC-10 Algorithm Description

The STU-III/LCT may incorporate the LPC-10(e) voice digitizer as defined in Appendix A. This section provides the description of the general Government standard LPC-10 algorithm, while the improvements embodied in LPC-10(e) are provided in Section 2.4.4.

The LPC-10 algorithm specifies both analysis of speech at the transmitter and synthesis of speech at the receiver.

Table 2-8. Standard Display Prompts for Required Conditions (Cont.)

Condition	Display Message
Receiving a Failed Call message	Secure Call Failed (20-char) Sec Call Failed (16-char)
Receiving a Release message	Secure Call Complete (20-char) Secure Complete (16-char)
Recognizing an AUTOVON preempt	Preempted (16- or 20-char)
Detecting a cryptoalarm	Hardware Failure (16- or 20-char)
CKL Checksum failure	Call KMC (16- or 20-char)

Note 1. An Abort message received from the KMC shall be treated as the condition "End of an unsuccessful KMC call".

LPC-10 models speech production as a linear, slowly varying filter excited by one of two sources; a random noise source for unvoiced sounds or a quasiperiodic pulse train for voiced sounds. The LPC-10 analyzer determines 53 bits of information characterizing each 22.5 ms frame of speech. A 54th bit, the alternating frame synchronization bit, is added for a data transmission rate of 2400 bits per second. A block diagram of the LPC-10 transmitter and receiver is provided in Figure 2-2.

2.4.3.1 Analyzer

The analog speech input to the analyzer is lowpass filtered and then sampled at a rate of 8 kHz by a μ-law CODEC device. The 8 bit μ-law speech sample is then converted to a linear sample of 12-bit magnitude. The speech samples are stored into both a pitch analysis buffer and a predictor coefficient analysis buffer at a frame rate of 44.4 frames per second. The analysis is performed to determine both the excitation signal parameters and the vocal tract filter parameters. The excitation signal is characterized by pitch and voicing parameters that are calculated during each speech frame. The vocal tract filter is characterized by RMS amplitude and reflection coefficients also calculated for each frame of speech.

A. Excitation Signal: The 180 speech samples of a 22.5 ms LPC frame from the pitch analysis buffer are low pass filtered for pitch and voicing extraction. This filtering removes the interference of high frequency spectral components in the pitch algorithm. The low pass, filtered signal is then fed to a second order inverse filter to spectrally flatten the signal and improve the pitch estimation. An Average Magnitude Difference Function (AMDF) pitch extraction is used for the raw pitch estimation. Voicing detection uses an energy measure, zero crossings analysis, and the AMDF maximum to minimum ratio to make the voicing decision. The voicing decision is made twice per frame, for a maximum of four voicing states: completely voiced, completely unvoiced, voiced to unvoiced transition, or unvoiced to voiced transition. The pitch and voicing results are smoothed to

Table 2-8. Standard Display Prompts for Required Conditions

Condition	Display Message
Far-end terminal on CKL (Type I calls only)	Far Key Compromised (20-char) Key Compromised (16-char)
Far-end terminal's key expired (Type I calls only)	Far Key Expired (16-or 20-char)
Far-end terminal's key is seed key	Far Key Invalid (20-char) Far Key Invalid (16-char)
Near-end terminal's key is seed key (optional)	1st line: Key Invalid (16-or 20-char) 2nd line: Call KMC to Init (16-or 20-char)
No common key exists	Incompatible key (16-or 20-char)
End of a successful KMC call (Type I calls only)	Key Updated (16-or 20-char)
End of an unsuccessful KMC call (Type I calls only)	Key Update Failure (20-char) Key Update Fail (16-char)
Receiving an Abort message [1]	Non-Secure Requested (20-char) Non-Sec Request (16-char)

Table 2-1 LPC-10(e) Frame Structure

Bit	Voiced	Nonvoiced	Bit	Voiced	Nonvoiced
1	RC(1)-0	RC(1)-0	28	RC(2)-4	RC(2)-4
2	RC(2)-0	RC(2)-0	29	RC(7)-0	RC(3)-5*
3	RC(3)-0	RC(3)-0	30	RC(8)-0	R-5*
4	P-0	P-0	31	P-4	P-4
5	R-0	R-0	32	RC(4)-4	RC(4)-4
6	RC(1)-1	RC(1)-1	33	RC(5)-0	RC(1)-5*
7	RC(2)-1	RC(2)-1	34	RC(6)-0	RC(2)-5*
8	RC(3)-1	RC(3)-1	35	RC(7)-1	RC(3)-6*
9	P-1	P-1	36	RC(10)-0	RC(4)-5*
10	R-1	R-1	37	RC(8)-1	R-6*
11	RC(1)-2	RC(1)-2	38	RC(5)-1	RC(1)-6*
12	RC(4)-0	RC(4)-0	39	RC(6)-1	RC(2)-6*
13	RC(3)-2	RC(3)-2	40	RC(7)2	RC(3)-7*
14	R-2	R-2	41	RC(9)-0	RC(4)-6*
15	P-2	P-2	42	P-5	P-5
16	RC(4)-1	RC(4)-1	43	RC(5)-2	RC(1)-7*
17	RC(1)-3	RC(1)-3	44	RC(6)-2	RC(2)-7*
18	RC(2)-2	RC(2)-2	45	RC(10)-1	Unused
19	RC(3)-3	RC(3)-3	46	RC(8)-2	R-7*
20	RC(4)-2	RC(4)-2	47	P-6	P-6
21	R-3	R-3	48	RC(9)-1	RC(4)-7*
22	RC(1)-4	RC(1)-4	49	RC(5)-3	RC(1)-8*
23	RC(2)-3	RC(2)-3	50	RC(6)-3	RC(2)-8*
24	RC(3)-4	RC(3)-4	51	RC(7)-3	RC(3)-8*
25	RC(4)-3	RC(4)-3	52	RC(9)-2	RC(4)-8*
26	R-4	R-4	53	RC(8)-3	R-8*
27	P-3	P-3	54	Sync	Sync

NOTES:
P = Pitch
R = RMS Amplitude
RC = Reflection Coefficient
* = Error Control Bit
Bit 0 = Least significant bit of data
Bit 5 = Least significant bit of error control
Order of transmission is from bit 1 to bit 54.
Nonvoiced = Unvoiced or In Voicing Transition

Table 2-7. Display Modes

Status Messages	Standard or vendor specific messages
Terminal ID and Authentication Messages	Far-end terminal authentication data (excluding the first eight characters and the special access field) shall be scrolled once during call setup. Far-end ID and authen-tication data (excluding the special access field in cer-tain cases) shall be rescroll-able upon appropriate request during a secure call. Near-end ID data shall be rescroll-able upon appropriate request when the CIK is inserted during nonsecure mode.
Security Classification	Throughout the secure portion of a call, the security classification (Table 2.9-3) and special identification or accesses shall be displayed continuously (with the exceptions specified in Section 2.9.2.2) on the first line of the display.
Compromise Information Message (Type I terminals only)	Any prompt message contained in the CIM shall be displayed (and automatically scrolled if necessary) to the user on the second line of the display during either the secure of non-secure mode. This display shall be provided only if the CIK is inserted. The STU-III design shall ensure that either the user is alerted at least once to the existence of the display message, or that the message is displayed until the user takes an action to clear the display message while the CIK is inserted.

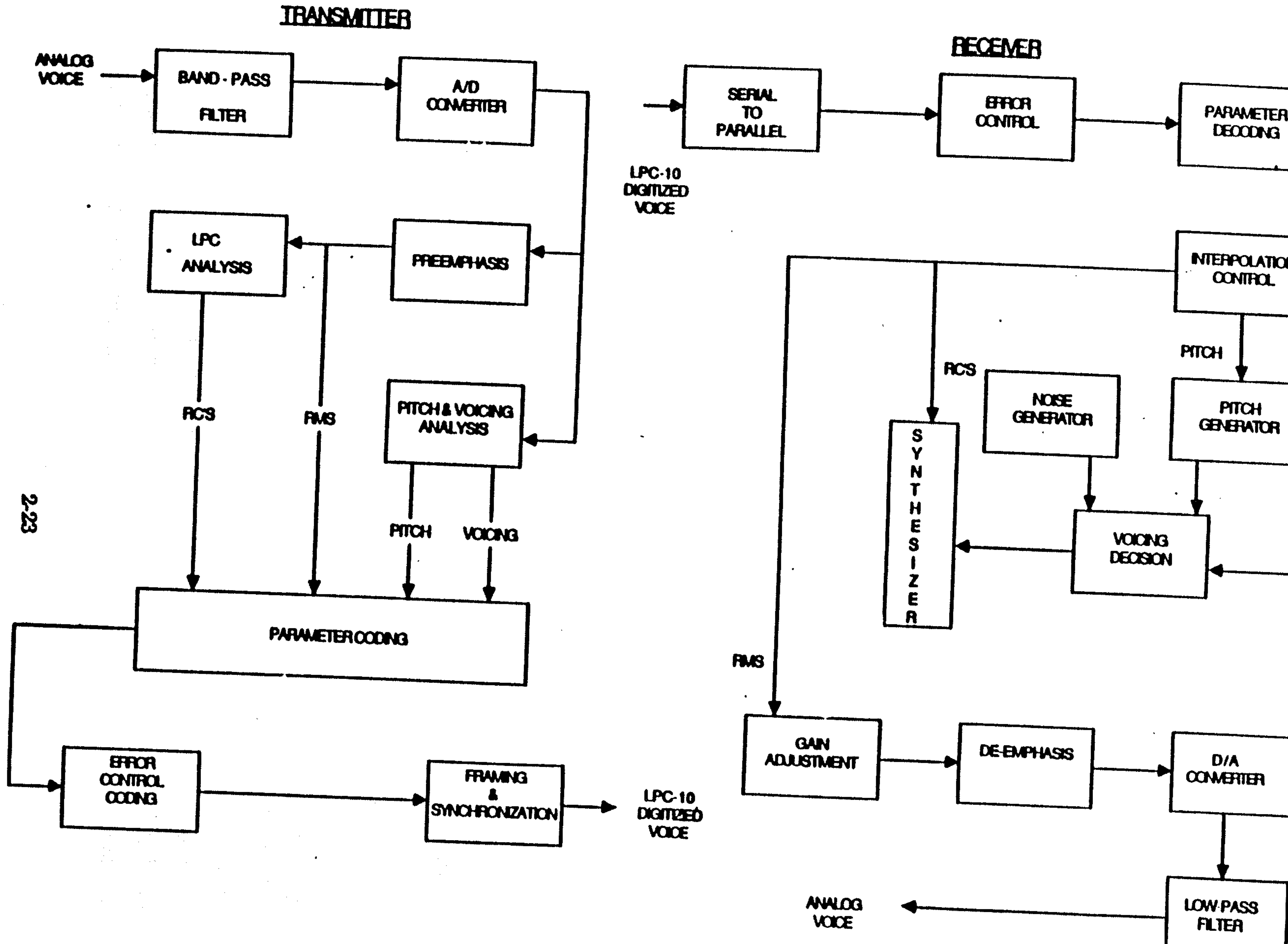

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Figure 2-2. LPC-10 Block Diagram

call status, terminal IDs, security level, and compromise information. The display modes are summarized in Table 2-7.

2.9.1 Status Message Display (MER)

The terminal display shall provide messages to assist the user with call setup and with failure situations. Table 2-8 contains the conditions for which standard messages are required. The STU-III shall display the strawman messages indicated in the table during those conditions. These messages may be displayed using any combination of upper case and lower case characters at the vendor's discretion. The terminal may also provide messages for conditions not listed in Table 2-8. The standard status messages appear on the first display line for one-line messages and on the first and second display lines for two-line messages. No standard status messages are used during secure STU-III to STU-III traffic modes since the classification must always occupy the top line of the display.

2.9.2 Security Classification and Special ID Field Display (MER)

2.9.2.1 Classification Display

During a secure call, the STU-III shall display the highest common classification of the two terminals, as determined during call set-up (Section 2.6.4.2), on the top line of the terminal display. The classification shall be displayed using one of two formats described in this section, full display of classification or truncation of classification. The STU-III shall truncate the length of the classification as shown in Table 2-9 if the special ID field of the far-end terminal is to be displayed (see section 2.9.2.2) and if the near-end STU-III display is less than 20 characters in length. The STU-III may either truncate the classification as described above or may display the full classification when the far-end terminal special field is to be displayed and the near-end STU-III display is 20 characters or more, or when the far-end terminal class is 4 or 5. All characters of the classification shall be displayed in upper case.

eliminate the effects of spectral power levels in the low pass, filtered waveform. The smoothing and track algorithm (DYPTRACK) employs two frames of internal delay.

B. Vocal Tract Filter: The speech samples from the predictor coefficient buffer are passed to a preemphasis filter to improve the stability of the vocal tract filter estimate. The spectral analysis is then performed semipitch synchronously for voiced frames. In this case, a 130-sample analysis window is aligned over the 180-sample speech frame such that it coincides synchronously with the pitch period. In semipitch-synchronous analysis, the spacing between successive analysis intervals is a multiple of the pitch period during voiced frames. During unvoiced frames, the analysis window is centered on the 22.5 ms frame. The 130 samples are then placed in a covariance matrix load array for processing.

The goal of the linear prediction algorithm is to minimize the mean-square error between the actual speech waveform and the predicted speech waveform. The solution to the covariance matrix yields the coefficients that minimize the mean-square prediction error. The covariance matrix elements are calculated from the 130 samples contained in the matrix load array after the removal of the short-term DC bias. A Cholesky decomposition algorithm is used to solve the matrix equations, yielding ten reflection coefficients. Reflection coefficients have the desirable property of being bounded between -1 and +1, assuring the stability of a synthesis filter. During the matrix calculation, an RMS energy parameter is calculated for the frame.

C. Parameter Encoding: The LPC parameters are encoded as specified in FED-STD-1015. (Section 2.4.2 of this document provides the relevant information.) When the frame is voiced, all ten reflection coefficients are encoded. When the frame is unvoiced, only the first four reflection coefficients are encoded since these coefficients are sufficient to represent unvoiced speech. In this case, a Hamming block code (8,4) is applied to

2.8.6 Multiline Version/Multiline Capability

The STU-III or a version thereof, with either internal or external circuitry, shall be compatible with Bell 1A2 Key Systems. The interface to the 1A2 system shall be via a standard 50 pin connector. The multiline version/capability shall accommodate a minimum of four TELCO circuits and shall support the "Hold" and "Intercom" capabilities provided by the host 1A2 keyphone controller.

2.8.7 Precedence/Routine Call Waiting Recommendations

It is recommended that the STU-III offer a capability to detect the presence of Precedence Call Waiting and Routine Call Waiting tones while in the secure mode. Upon detecting these tones, the STU-III shall provide a momentary aural indication, interrupt the normal display on the second line and display "Call Waiting" for approximately 2 seconds, and then return to the normal display. These indications shall be repeated each time the Call Waiting Tones are presented to the STU-III.

2.8.7.1 Precedence Call Waiting

A Precedence Call Waiting tone will consist of 3 bursts of a 440Hz tone. Each burst is ON for 100ms ± 20ms and OFF for 20ms ± 10ms. The Precedence Call Waiting tone may be repeated every 10 seconds.

2.8.7.2 Routine Call Waiting

A Routine Call Waiting tone consists of one burst of 440 Hz tone which is ON for a duration of 300ms ± 50ms.

2.9 USER INTERFACE (MER)

The STU-III shall provide an alphanumeric display to present information to the user. The display shall provide a minimum of two lines with a minimum of 16 characters for each line. The display shall provide messages concerning protect the four most significant bits (MSBs) of RMS amplitude and the four MSBs of the four reflection coefficients.

2.4.3.2 Synthesizer

At the receiver, frame synchronization is acquired. The received LPC parameters are interpolated pitch synchronously and allocated to either an excitation algorithm or to a synthesis filter for speech production. Pitch and voicing information bits are decoded as indicated in FED-STD-1015. The voicing decision is decoded on a half frame basis depending on the received voicing parameters and the channel conditions. The pitch period, RMS amplitude, and either four or ten reflection coefficients are decoded, dequantized and buffered. When the frame is unvoiced, the channel errors are corrected prior to the parameters being decoded.

Voiced speech is synthesized with a tenth-order, all-pole filter excited by pitch-synchronous pulses. Unvoiced speech is synthesized with a fourth-order, all-pole filter with pseudo-random or quasi-random excitation. Speech in voicing transition is synthesized in the same way as for voiced speech during the voiced part of the frame, and in the same way as for unvoiced speech during the unvoiced part of the frame. The synthesized samples are scaled to match the RMS amplitude from the transmitter for every pitch epoch.

The resulting digital waveform is passed through a deemphasis filter to compensate for the preemphasis filtering of the transmitter. The deemphasized speech samples of this waveform are compressed before being output to a μ-law CODEC. The output of a CODEC is fed to the analog audio subsystem (see section 2.3) for final processing.

2.4.4 LPC-10(e) Improvements

2.4.4.1 Analysis

2.8.5.1 Digital Network Requirements

(TBSL)

2.8.5.2 Digital Network Recommendations

(TBSL)

The enhancements to the DoD standard LPC-10 analysis are as follows:

- New preemphasis/deemphasis algorithm that boosts high frequency spectral components by 6 dB/octave from 2 kHz to 4 kHz in order to enhance high band analysis of low spectral components.

- Adaptive placement of the analysis window at speech onsets in order to capture the abrupt voice onsets of both voiced and unvoiced consonants. The method performs a single-order prediction in real time from the preemphasized speech waveform. The difference of the forward prediction coefficients filtered over 16 samples determines the onsets.

- The low pass filter used for pitch and voicing extraction is a 19-tap finite impulse response (FIR) low pass filter.

- The analysis is scaled for S12 input level.

2.4.4.2 Synthesis

The enhancements to the LPC-10 synthesis are as follows:

- The excitation signal for voiced sounds is supplemented with a random noise source to recreate the variations of pulse shape and period normally found in voiced speech.

- The excitation signal for unvoiced sounds is supplemented with random pulses to improve the synthesis of spiky and abrupt plosive sounds.

- The reflection coefficients are fed to the synthesizer twice, with the output of the first pass through the synthesizer used as the input to the second pass through the same synthesizer routine. This is equivalent to creating a 20-tap synthesis filter instead of a 10-tap synthesis filter, shaping the LPC spectral envelope and improving the perceptual voice quality.

Several of the STU-III terminal designs will be capable of being configured for operation on a mobile cellular radio telephone network. These terminals shall be directly interoperable with all STU-IIIs connected to the standard telephone network.

2.8.4.1 Cellular Network Requirements

The signaling requirements for cellular operation are provided in FSVS-210.

2.8.4.2 Cellular Network Recommendations

A mobile version of the STU-III should, as a minimum, be vehicularly mounted and powered, and operate over cellular and mobile radio telephone circuits.

2.8.5 Digital Network (56/64 Kbps)

An architecture which would support a transition to wideband telecommunication systems would be useful. Characteristics of the terminal would include:

- Operation with AT&T's Circuit Switched Digital Capability (ISDN) and other future wideband voice and data systems. Concurrent operation in more than one wideband system is not envisioned.

- Full duplex encrypted voice and data at rates up to 64 kbs. Either CVSD, mu-law PCM, or Adaptive Differential PCM would be used for voice modes.

- Data and BLACK I/O via EIA RS-449 interfaces.

- Key Generator and Key Management functions.

- Modularity of RED & BLACK I/O to provide capability of interfacing to a wide range of future unspecified networks.

- The synthesizer is scaled for S12 input level.

- The spectrum coefficients are interpolated every pitch period.

2.4.4.3 Excitation

- A new pattern matching algorithm is used for voicing.

2.5 DATA PROCESSING

This section addresses the STU-III required synchronous and recommended asynchronous secure data modes.

2.5.1 2400 Synchronous Data Mode (MER)

The STU-III/LCT shall be capable of transmitting synchronous data securely at 2400 bps as a MER, using the signaling protocols specified in FSVS-210. Any vendor's Type I or Type II STU-III/LCT shall be interoperable with any other LCT in this mode.

2.5.1.1 General Mode Characteristics (MER)

The STU-III initiates the data mode signaling protocols in the full duplex transmission mode when (1) the STU-III Data Mode control is activated by the user, and (2) the STU-III detects the Terminal Ready signal from the DTE, indicating the data device is active. Transmission of data is controlled by the STU-III/DTE interface signals as specified in Section 2.5.1.2. FSVS-210 specifies the signaling protocol requirements for the STU-III/STU-III interaction.

The STU-III initiates the data mode signaling protocols in the half duplex transmission mode when (1) the data device is connected to the data port of the STU-III, (2) the STU-III Data Mode control is activated by the user, and (3) the

2.8.2 AUTOVON/Defense Switched Network (DSN)

As a minimum the STU-III shall interface to AUTOVON 4-, 6-, and 8-wire system using an interface equipment equivalent to the Information Development and Applications, Inc. (IDEAS) AUTOVON Line Interface Unit, V-5891. The interface between the STU-III and ALIU shall be on a 4-wire basis, using Tx pair and Rx pair for all signalling, supervision, and plain/secure communications. Additional STU-III capabilities which permit direct connection to 4-w AUTOVON facilities without the need for an ALIU shall also be permitted.

Note: See System Manager paper, Technical Description of IDEAS ALIU, for interface information.

2.8.3 Foreign Country Networks

A single version of the STU-III (the "Overseas Version") shall be capable of operating over telecommunications systems in the United Kingdom (UK), Federal Republic of Germany (FRG), and Belgium (termed "host Nations") and shall be designed to achieve Connection Approval (CA) in these countries. This single version shall either meet all electrical, mechanical and safety requirements with circuitry internal to the equipment, or it may use external modules to meet the various Host Nation CA requirements. In addition to satisfying the above CA requirements, the power supply in the Overseas Version STU-III also shall meet the requirements of paragraph 3.2.1.2.

As a recommendation, modular BLACK I/O would be useful for interconnection to a variety of foreign systems. This module could be tailored to the unique interconnect requirements of each host nation. The module would also have programmable features that would delay the transmission of answer tones or other nonvoice signals for a period of time from 0 to 5 seconds in 1 second increments after the terminal goes off-hook to the network.

2.8.4 Cellular Network

data device indicates that data is available for transmission as specified in Section 2.5.1.2.

The STU-III shall incorporate the signaling structure for the 2400 bps synchronous data as specified in FSVS-210. This mode shall put no restrictions on the format or content of the transmission information from the data device. The STU-III shall be capable of performing a resync operation when an OOS indication from the data device (or from some external source) is detected. The STU-III shall transmit all data prior to a mode change out of the data mode. The STU-III may append up to 127 bits of additional data (set to all ones) after the RS signal goes low.

2.5.1.2 DTE Interface Signal Description (MER)

The STU-III/LCT shall provide a data device interface compatible with EIA Standard RS-449 or RS-232-C incorporating the mandatory signals shown in Figure 2-3 and listed below.

Guidelines for interconnection between interface circuits using RS-449 and RS-232-C is specified in EIA Bulletin No. 12.

A. TR (TERMINAL READY): DTE-to-STU-III

Terminal Ready (TR) indicates the DTE is in a mode where it can send or accept data; the DTE can assert this signal at any time, even if the STU-III is in the idle or voice mode.

B. DM (DATA MODE): STU-III-to-DTE

Indicates that a non-idle STU-III has selected the data mode. The STU-III with DM asserted in half duplex mode may prompt user to enable DTE if TR is not asserted. No data mode operation can occur until both TR and DM are asserted concurrently. The local DTE may flash its TR signal (ON-OFF-and EMV lines with or without near-end echo. With far-end and near-end echo present the STU-III shall operate at $1x10^{-3}$ or better on the CMD, CMV and CPD lines.

Each of these bit error rate requirements shall be met independently on each line identified.

C. Cellular Radio Operations: The requirements for the STU-III operating with the V.32 modem are identical to those stated for operation with the V.26 modem in Section 2.7.1.2, Paragraph D with 2 exceptions: (1) The differences in channel conditions stated in Section 2.7.2.4, and (2) No time limit is specified for the restoration of the bit error rate after the cellular impairment when operating at 9600 bps.

2.8 NETWORK INTERFACE

2.8.1 DDD Network

The STU-III terminals shall provide for a direct subscriber connection to the Direct Distance Dialing network (DDD).

2.8.1.1 DDD Network Interface Requirements (MER)

The terminals shall accommodate a 2-wire commercial interface with typical DC signaling and supervision. The physical DDD-interconnection requirement for the STU-III is specified in Section 3.1.1.4. The STU-III shall be capable of supporting both DTMF and dial pulse dialing. The terminals shall be designed and registered with the FCC in accordance with Part 68 (Interconnect) of the FCC Rules and Regulations.

2.8.1.2 DDD Network Interface Recommendations

It is recommended that performance for interfacing and connecting with the DDD network conform to EIA RS-470 specification.

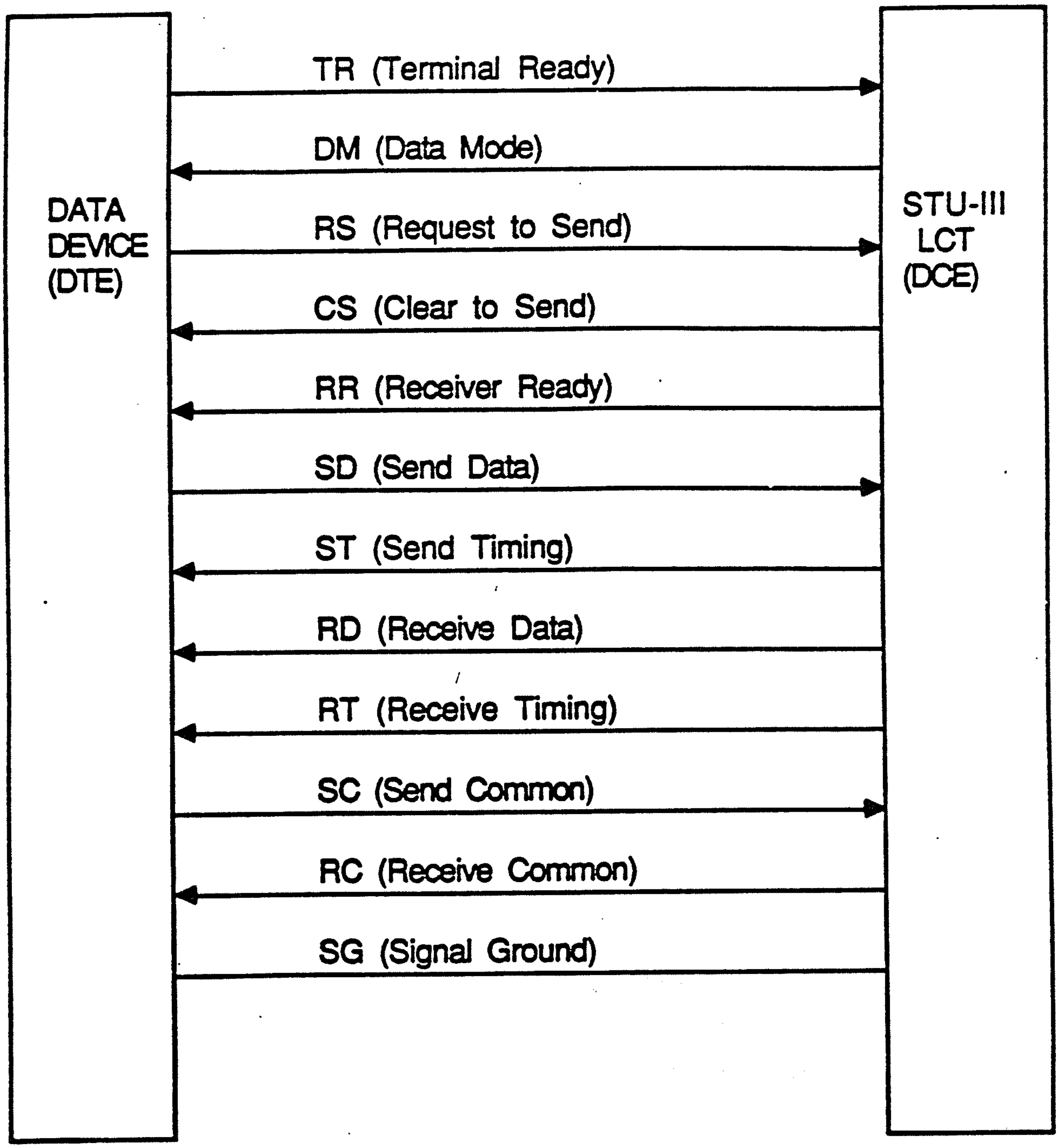

Figure 2-3. EIA-RS-449/232-C Interface

The STU-III V.32 modem shall be designed to maintain synchronization 100 percent of the time after acquisition for each of the channel conditions associated with an initial synchronization requirement. Demonstration tests to verify the ability to retain synchronization shall be conducted for a minimum of 10 minutes.

In addition to meeting the requirements for synchronization at 4800 and 9600 bps with and without the retransmission protocol specified above, the STU-III starting full duplex call setup in the V.32 mode shall meet the initial synchronization requirements specified in Table 2-4 for the 2400 bps modem when fallbacks and synchronization at 2400 bps are included in the count of successful synchronizations. Consistent with the requirements specified for 2400 bps, this requirement shall be met independently for each of the eight Government lines, with ADPCM, and with or without near-end and far-end echo. In meeting this requirement, a full duplex call setup attempt starting at 4800 or 9600 bps is regarded as successful if it synchronizes into the data rate selected by the user before call setup is attempted, achieves call setup at a lower rate, or falls back in accordance with the Call Setup Failure Fallback signaling specified in FSVS-210, Section 2.2.3.1.6 and achieves synchronization at 2400 bps.

[The requirement stated in this paragraph applies only to the full duplex mode of operation.]

B. Bit Error Rate Performance: At 4800 bps, the STU-III modem shall operate at a bit error rate of $5x10^{-4}$ or better on the CMD, CMV, CPD, EMD and EMV lines with or without near-end and far-end echo. On the CPV, EPD and EPV lines the BER requirement at 4800 bps is reduced to $1x10^{-3}$ or better without echo and $5x10^{-3}$ or better with near-end echo. No requirement is imposed at 4800 bps on these three lines when far-end echo is present.

At 9600 bps, the STU-III modem shall operate at a bit error rate of $5x10^{-4}$ or better on the CMD and CMV lines and $5x10^{-3}$ or better on the CPD, EMD

ON) when it wants the remote terminal to send a new crypto synchronization (i.e., current received data is out-of-sync).

C. RS (REQUEST TO SEND): DTE-to-STU-III

DTE should not assert and STU-III shall not accept this signal unless both TR and DM are asserted. DTE asserts RS when it has data to send indicating data available to the STU-III for transmission as defined in FSVS-210 and awaits CS. DTE removes signal when it has no data to send and should not send data (puts MARKS on SD line) after it removes RS, even if CS remains asserted. Removal of the RS signal indicates to the STU-III that data is no longer available. The STU-III with a half duplex DTE operating on a full duplex link shall initiate one way cryptosync when RS is removed (i.e., data is no longer available for transmission by the STU-III) and shall send a START and initiate data transfer in one direction when a valid RS is received from the DTE (i.e., data is again available to the STU-III for transmission). STU-III shall enter another mode or exit the secure call after RS is removed, but not before it has encrypted and transmitted all data (SD) that it accepted before RS was dropped.

D. CS (CLEAR-TO-SEND): STU-III-to-DTE

STU-III shall assert CS when it can accept SD from DTE such that the signaling requirements of FSVS-210 are satisfied. STU-III can assert CS in anticipation of cryptosync completion if it buffers all data it accepts after asserting CS and delivers all buffered data to the channel after DTE removes RS. The STU-III shall not remove CS in response to the DTE removing RS until all data accepted while RS was asserted has been delivered and the STU-III is ready to accept a new RS. Likewise, the DTE should not assert a new RS until the STU-III removes CS.

Table 2-6. Performance Requirements for 9600 bps Modems

Transmission Channel Conditions						Requirements for V.32 Modem Operation at 9600 bps
Government Lines	ADPCM (32 kbps)	Near-End Echo	Far-End Echo	Cellular Impairment	Received Signal Level	
CMD,CMV, CPD,EMD, EMV	—	—	—	—	-9 to -43 dBm	-Initial Synchronization ≥ 90% (93%) -Retain Synchronization > 10 min -BER ≤ 5 x 10^{-4} (CMD, CMV) -BER ≤ 5 x 10^{-3} (CPD,EMD,EMV)
CMD,CMV	1 Hop	—	—	—	-9 to -35 dBm	-Initial Synchronization ≥ 90% (93%) -Retain Synchronization > 10 min -BER ≤ 5 x 10^{-4}.
CMD,CMV, CPD,EMD, EMV	—	E ≤ -12 dBm D ≤ 12 ms	—	—	-9 to -43 dBm	-Initial Synchronization ≥ 90% (93%) -Retain Synchronization > 10 min -BER ≤ 5 x 10^{-4} (CMD, CMV) -BER ≤ 5 x 10^{-3} (CPD,EMD,EMV)
CMD,CMV	1 Hop	E ≤ -12 dBm D ≤ 12 ms	—	—	-9 to -35 dBm	-Initial Synchronization ≥ 90% (93%) -Retain Synchronization > 10 min -BER ≤ 5 x 10^{-4}.
CMD,CMV, CPD (With no frequency offset)	—	E ≤ -12 dBm D ≤ 12 ms	S/E ≥ 0 dB D ≤ 750 ms	—	-15 to -36 dBm	-Initial Synchronization ≥ 75% (83%) -Retain Synchronization > 10 min -BER ≤ 1x 10^{-3}
CMV	—	—	—	≤ 5 sec fade or dropout	-9 to -43 dBm	-Retain System Synchronization.
CMV	—	—	—	up to ± 100 microsecond instantaneous change in path delay	-9 to -43 dBm	-Retain System Synchronization.

at least 93 percent of the attempts under the same conditions, but 90 percent of the attempts shall succeed without the use of the retransmission protocol. In computing these success rates, a call setup attempt is regarded as successful if it achieves synchronization into the data rate initially selected. A shift down in rate, a Failed Call restart at 2400 bps, and a Failed Call shall be regarded as failures.

E. RR (RECEIVER READY): STU-III-to-DTE

STU-III shall assert RR when it is delivering valid RD to the DTE; when RR is not asserted, the RD line shall be constant MARKs. The STU-III shall assert RR when it delivers the first valid decrypted data traffic bit to the DTE and shall remove it upon detection of Escape after it has delivered the last valid data bit prior to the Escape. Note that this implies that the STU-III shall buffer enough receive data to accommodate the detection of Escape.

F. SD (SEND DATA): DTE-to-STU-III

Data to be encrypted and transmitted by STU-III; synchronous with ST. SD should be constant MARKs when any one of the signals TR, DM, RS, CS is not asserted.

G. ST (SEND TIMING): STU-III-to-DTE

Clock for SD. The STU-III shall assert CS such that the first rising edge of ST after DM, RS, CS and TR signals are high shall frame the first bit to be encrypted. The III shall sample SD on the falling edge of ST. DTE should remove RS on the rising edge of ST.

H. RD (RECEIVE DATA): STU-III-to-DTE

Traffic Data decrypted by STU-III; synchronous with RT. RD shall be constant MARKs when any one of the signals TR, DM, and RR is not asserted. RD shall be framed by the rising edges of RT.

I. RT (RECEIVE TIMING): STU-III-to-DTE

Clock for RD. The STU-III shall assert RR such that the first rising edge of RT after DM, TR and RR signals are high shall frame the first decrypted bit. The STU-III shall frame RD with the rising edges of RT.

Table 2-5. Performance Requirements for 4800 bps Modems

Transmission Channel Conditions						Requirements for V.32 Modem Operation at 4800 bps
Government Lines	ADPCM (32 kbps)	Near-End Echo	Far-End Echo	Cellular Impairment	Received Signal Level	
CMD,CMV, CPD,EMD, EMV	4 Hops	—	—	—	-9 to -43 dBm	-Initial Synchronization ≥ 90% (93%) -Retain Synchronization > 10 min -BER ≤ 5 x 10^{-4}
CPV,EPD, EPV	—	—	—	—	-9 to -43 dBm	-Initial Synchronization ≥ 70% (79%) -Retain Synchronization > 10 min -BER ≤ 1 x 10^{-3}
CMD,CMV, CPD,EMD, EMV	4 Hops	E ≤ -12 dBm D ≤ 12 ms	—	—	-9 to -43 dBm	-Initial Synchronization ≥ 90% (93%) -Retain Synchronization > 10 min -BER ≤ 5 x 10^{-4}
CPV,EPD, EPV	—	E ≤ -12 dBm D ≤ 12 ms	—	—	-9 to -43 dBm	-Initial Synchronization ≥ 70% (79%) -Retain Synchronization > 10 min -BER ≤ 5 x 10^{-3}
CMD,CMV, CPD,EMD, EMV (With no frequency offset)	—	E ≤ -12 dBm D ≤ 12 ms	S/E ≥ 0 dB D ≤ 750 ms	—	-15 to -36 dBm	-Initial Synchronization ≥ 75% (83%) -Retain Synchronization > 10 min -BER ≤ 5 x 10^{-4}
CMD	—	—	—	≤ 5 sec fade or dropout	-9 to -43 dBm	-Retain System Synchronization -Restore BER ≤ 5 x 10^{-4} within 100 ms after dropout
CMD	—	—	—	up to ± 100 microsecond instantaneous change in path delay	-9 to -43 dBm	-Retain System Synchronization -Restore BER ≤ 5 x 10^{-4} within 100 ms after change

2.5.1.3 Data Mode Interface Characteristics

The STU-III shall sample the SD line on the falling edge of ST and frame the individual bits by the rising edge of ST. During the 2400 bps synchronous data operation, no data bits shall be lost in the encryption/decryption process (i.e., the first bit following start shall be the first bit of data encrypted by the first bit of key). The first rising edge of the ST signal when DM, RS, CS and TR are high frames shall be the first bit to be encrypted (bit 1). On the first falling edge of the ST signal, the STU-III shall strobe in the first bit for the encryption process. On the first rising edge of the RT signal, the STU-III shall frame the first decrypted bit when DM, TR and RR signals are high. Neither the encryption nor the decryption process shall introduce inversion in the plaintext data stream. The STU-III shall be capable of accepting an out-of-sync indication from an external device, or from the user, which will cause a resync sequence.

2.5.1.4 Data Mode Interface Recommendation

It is recommended that the out-of-sync indication be provided by the data terminal via transitioning the TERMINAL READY (TR) signal to the STU-III/LCT from a 1 to 0 and back to 1 within a maximum of 100 msec., with the zero state maintained for a minimum of 20 msec.

2.5.2 Asynchronous Full Duplex Data Mode

The STU-III may, as an option, provide full duplex, asynchronous data modes to the users. The asynchronous data modes could provide a communication link between a Micro-KMODC and a LIT as one anticipated application. Three full-duplex, asynchronous data modes are specified below: 1200 bps half-rate, 1200 bps full-rate and 2400 bps full-rate. If the STU-III implements one of the specified interoperable asynchronous full duplex data modes, then the design of theasynchronous data mode shall comply with the requirements presented in this section.

While operating on signals received over a variety of transmission channels, the STU-III operating with the V.32 modem shall meet performance requirements on initial and long term system synchronization, bit error rates, and recovery after interruptions in a cellular environment. Tables 2-5 and 2-6 specify the channel characteristics for the 4800 bps and 9600 bps modes respectively and identify the requirements that must be met when operating over the specified channels. Further details of the channel characteristics identified in Tables 2-5 and 2-6 are specified in Sections 2.7.1.1 and 2.7.2.4, while details of the performance requirements are specified below. These performance requirements apply to both the full and half duplex modes of operation and shall be met independently by each mode implemented in the STU-III.

A. Synchronization: After a secure call is initiated, the STU-III shall acquire initial system synchronization at the success rates specified in Tables 2-5 and 2-6 for the various channel conditions identified. STU-IIIs which have not implemented the retransmission protocol specified in FSVS-210, Section 2.2.1.3.1 shall achieve the synchronization rates specified by the first number in the table. STU-IIIs which have implemented the retransmission protocol shall achieve the synchronization rates specified by the second number in the table. In addition, however, such terminals shall achieve the rates specified by the first number in the table without the use of the retransmission protocol. The success rates shall be computed for the groups of lines specified by averaging the rates of synchronization over all of the lines in the group. Thus, for example, a STU-III without the retransmission protocol shall achieve initial system synchronization into a 4800 bps traffic mode at an average rate of at least 90 percent of the secure call setup attempts when tested over the five lines: CMD, CMV, CPD, EMD, and EMV, with no echo. If the STU-III has the retransmission protocol implemented, it shall acquire system synchronization at an average rate of

2.5.2.1 1200 bps Half-Rate Asynchronous Data Mode

The interface between the STU-III and the data device shall be a 1200 bps asynchronous interface. The STU-III/STU-III interaction shall be 2400 bps synchronous full duplex, half-rate (effective data rate is 1200 bps).

The Asynchronous Data function is specifically broken into three separate levels: the asynchronous red data port; the asynchronous-to-synchronous format conversion; and the black side transmission. The remainder of this section will specify the requirements of each level.

2.5.2.1.1 Red Asynchronous Data Port. The terminal asynchronous data interface shall support a 10-bit character format, including 1 start bit, 8 data bits, and 1 stop bit. The recommended interface shall support a nominal 1200 bps rate or 2400 bps rate +1%/-2.5%, depending on the mode selected. Data characters shall be exchanged compatible with EIA RS-449 standards. The RS-449 interface signals are specified in Section 2.5.1.2.

2.5.2.1.2 Format Conversion. The STU-III/LCT terminal shall provide a format conversion mechanism which converts asynchronous data from the red data port to a synchronous bit stream for black side processing. This conversion specification is based on the CCITT V.26 ter recommendations.

In the transmitting STU-III, the format converter shall accept a data stream of asynchronous characters from the data device at a nominal rate of 1200 bits per second. The asynchronous data shall be converted to a form suitable for transmission synchronously at 1200 bits per second +/- 0.01%. The converter shall derive its transmitting clocks from an internal clock source or, alternatively, from the asynchronous data received from the local DTE. The converter shall be capable of transmitting characters contiguously or with an additional arbitrary integer number of stop bits between characters.

retraining from any of the other modulation formats incorporated in the terminal.

2.7.2.3 Requirements on Half Duplex "V.32" Option

2.7.2.3.1 Call Setup Option. STU-IIIs offering the "V.32" half duplex option may offer the capability to enter the mode through initial call setup in accordance with the provisions of FSVS-210.

2.7.2.3.2 Retrain Requirement. STU-IIIs offering the 4800 bps or 9600 bps half duplex option shall incorporate the ability to enter the mode through retrain signaling as specified in FSVS-210. This shall include the ability to enter the mode from any of the other modulation formats included in the terminal.

2.7.2.4 Channel Characteristics for V.32 Modem Testing

The characteristics of the simulated channels used to specify the requirements for the V.32 modems operating at 4800 and 9600 bits per second are the same as those specified for the V.26 modem in Section 2.7.1.1 with four exceptions: (1) The impairments due to ADPCM will be used at 4800 and 9600 bps only selectively, (2) The signal to far-end echo (S/E) ratio specified is increased to greater than or equal to 0 dB when testing over the channel with far-end echo, (3) The specification for instantaneous change in path delay is reduced to changes up to ± 100 microseconds when testing performance in the cellular radio environment, and (4) The standard range used for received signal levels is reduced to -9 to -43 dBm when testing for compliance to the performance requirements. These changes are identified in Tables 2-5 and 2-6 which specify the channel conditions for testing the 4800 bps and 9600 bps modems respectively.

2.7.2.5 Performance Requirements for V.32 Modems

The format conversion of the asynchronous data mode shall adhere to the specific characteristics discussed below:

A. Character Format: The character format selected shall be the same for both the transmitting STU-III and receiving STU-III. The converter shall accept a 10 bit character format (a start bit, followed by 8 data bits, and a stop bit). If the option for interfacing with additional character formats is selected (e.g. 6-bit data, or 7-bit data), it is acceptable that the data bits be replaced by additional stop bits. In this situation, the STU-III terminal shall provide a manual switch for option selection, and it will be the responsibility of the users to ensure that a compatible option is selected.

B. Intracharacter Transmit Signal Rate: The intracharacter signaling rate (signaling rate of the start bit and data bits within each character) provided by the data device must be 1200 bps, +1%/-2.5%.

C. Character Interval: The character interval (the time interval between successive start bits) provided by the data device must be greater than 8.25 ms.

D. Rate Adjustment: The converter shall, as often as is necessary, not transmit the stop bits of incoming characters in order to attain the nominal transmission rate. No more than one stop bit shall be deleted for any eight consecutive characters. If any stop bits are deleted, the receiving STU-III shall restore the stop bits before outputting the characters to the data device. When the character rate provided by the data device is less than 120 characters per second, the converter shall therefore insert extra stop bits in between the transmitted characters.

E. Transmit Break Signal: Upon receiving break signals (at least 23 bits all of "start" polarity) from the data device, the STU-III shall not do anything other than faithfully transmit the break signals to the far-end terminal.

2.7.2.1.2 Retrain Request. STU-IIIs transmitting CAP Version No. 0001 in accordance with the previous paragraph shall include the capability to interpret and respond to the Retrain Request Message in any of the full or half duplex modulation formats incorporated in the terminal. For this requirement the response is defined as the Retrain ACK or Retrain NACK as specified by FSVS-210. The ability to retrain into one of the defined modes shall be indicated by the bits set in the CAP segment. This provision specifically allows a terminal with only the 2400 bps rate to incorporate the retrain feature between full and half duplex. This would be indicated by transmitting CAP Version No. 0001 with only Bits 70 and 71 set in the transmission capabilities segment of the CAP message.

2.7.2.1.3 ESCD Requirement. STU-IIIs transmitting CAP Version No. 0001 shall transmit ESCD in both the 2400 and V.32 modes wherever ESD/ESCD is otherwise shown as a switch selectable option. Such terminals shall, however, retain the capability to respond correctly to the ESD tone. The switch selectable requirement is eliminated from such terminals.

2.7.2.2 Requirements on the Full Duplex V.32 Option

2.7.2.2.1 Call Setup Option. STU-IIIs offering the V.32 full duplex option may offer the capability to enter the mode through initial call setup in accordance with the provisions of FSVS-210. If this option is implemented, it shall include the ability to perform call setup at the 4800 bps transmission rate.

2.7.2.2.2 Restart Requirement. STU-IIIs offering the ability to perform initial call setup in a V.32 mode shall incorporate the Failed Call Restart capability to restart call setup automatically in the 2400 bps V.26 mode as defined in FSVS-210.

2.7.2.2.3 Retrain Requirement. STU-IIIs offering a V.32 full duplex option shall incorporate the ability to enter the mode through retrain signaling as specified in FSVS-210. This shall include the ability to enter the mode by

F. Intracharacter Receive Signal Rate: In the receiving LCT, the intracharacter signaling rate output to its DTE shall be in the range of 1200 to 1227 bps. The nominal length of the start and data bits for all characters shall be the same. The number of stop bits shall not be reduced by more than the inserted extra stop bits from the transmitting STU-III to allow for over-speed in the transmitting STU-III.

G. Receive Break Signal: The break signals (at least 23 bits all of "start" polarity) received from the transmitting STU-III shall be transmitted to the data device. The DTE will transmit at least 20 bits of "stop" polarity after the break signal before sending further data characters. The format converter shall regain character synchronization from the first "stop" to "start" transition following the break signal and the stop bits.

2.5.2.1.3 Black Side Data Rate Transmission Format. The STU-III terminal shall transmit the asynchronous data to the far-end terminal in a 2400 bps synchronous transmission. There shall be no data bits lost in the encryption/decryption process (i.e., the first bit out of the data device shall be the first bit encrypted by the first bit of key, and shall be the first bit transmitted after START; there shall be no sacrifice bits in the data mode caused by the encryption process). The black side of the STU-III transmission shall incorporate a half-rate BCH coded transmission; thus the information throughput rate is a nominal 1200 bps. The BCH block encoder may cause the data bits of characters to be split between blocks. A BCH data block may contain padding for up to 127 bits in the event the data has one stray bit for the final block to be transmitted. The BCH format and the overall transmission signaling shall be as defined in FSVS-210.

2.5.2.2 1200 bps Full-Rate Asynchronous Data Mode

(TBSL)

2.5.2.3 2400 bps Full-Rate Asynchronous Data Mode

Signaling Plan (FSVS-210). When the strap is in the ESCD position, the modem shall transmit the ESCD tone in accordance with the Signaling Plan. The selection "strap" may be implemented in either hardware or software as an implementation option. The strap shall be implemented, however, in such a way that, once set, the selection shall remain in force through normal use and power outages unless explicit steps are taken to change the selection.

2.7.2 V.32 Modem (Option)

As an option STU-IIIs may incorporate an interoperable V.32 modem for full or half duplex communication.

When a V.32 interoperable modem is incorporated in the STU-III, it shall include features in accordance with this section of the specification. Signaling for the V.32 mode shall be in accordance with the V.32 sections of FSVS-210.

As specified in FSVS-210, the V.32 full duplex modulation format at 4800 and 9600 bps is specified by the CCITT V.32 standard.

STU-IIIs may incorporate the V.32 full duplex mode or the "V.32" half duplex mode individually or may incorporate both the full and half duplex modes together. In either mode, the STU-IIIs may incorporate the 4800 bps rate, the 9600 bps rate or both.

2.7.2.1 Requirements on all V.32 Options

2.7.2.1.1 CAP Version No. 0001. Any STU-III incorporating either a full or half duplex V.32 mode shall transmit the CAP segment of the CAP/SV message in accordance with the specification for CAP Version No. 0001. When transmitting the CAP segment, the STU-III shall set the bits in the CAP segment to indicate its own capabilities to the STU-III at the remote end.

In this mode, the interface between the STU-III and the data device shall be a 2400 bps asynchronous interface. The STU-III/STU-III interaction should be full duplex, 2400 bps full-rate.

2.5.2.3.1 Red Asynchronous Data Port. Same as Section 2.5.2.1.1.

2.5.2.3.2 Format Conversion. The conversion specification is based on the CCITT V.26 ter recommendations.

In the transmitting LCT, the format converter shall accept a data stream of asynchronous characters from the data device at a nominal rate of 2400 bits per second. The asynchronous data shall be converted to a form suitable for transmission synchronously at 2400 bits per second +/- 0.01%. The converter shall derive its transmitting clocks from an internal clock source or, alternatively, from the local DTE. The converter shall be capable of transmitting characters contiguously or with an additional arbitrary integer number of stop bits between characters.

A. Character Format: Same as in Section 2.5.2.1.2.

B. Intracharacter Transmit Signal Rate: The intracharacter signaling rate provided by the data device must be 2400 bps, +1%/-2.5%.

C. Character Interval: The character interval provided by the data device must be greater than 4.125 ms.

D. Rate Adjustment: Same as in Section 2.5.2.1.2, except when the character rate provided by the data device is less than 240 characters per second. The converter shall insert extra stop bits between the transmitted characters.

E. Transmit Break Signal: Same as in section 2.5.2.1.2.

(3) ESD level (2100 Hz)
0 dBm/600 ohm ± 3 dBm.

The capability should be provided to attenuate the transmitted signals from 0 dB to 12 dB in 1 dB increments while maintaining their relative levels when measured at the telephone interface.

E. Receive Signal Level: The STU-III terminal should also provide the capability to amplify or attenuate the received signals in order to optimize the performance of the modem receiver.

F. Crosstalk: When multiple handsets are used, the crosstalk from the modem transmit output to the modem receive input should be at least 45 dB down relative to a 1004 Hz transmit signal at 0 dBm in a 4-wire, full duplex operation. The transhybrid loss in a 4-wire to 2-wire converter should be at least 30 dB.

G. Loopback (Analog and Digital): The modem transmit output should be capable of being looped back to the modem receive input on the terminal side of the hybrid. The gain of this path should be 0 dB ±3 dB at 1004 Hz. When disabled, the gain of this path should drop by 60 dB minimum. The modem receive input should be capable of being digitally looped back to the modem transmit output. The gain of this loopback path should be 0 dB at 1004 Hz. (Note: Normally the modem scrambles transmit traffic with one scrambler and descrambles receive traffic with the other scrambler. For the purpose of implementing loopback, the STU-III must use the same scrambler on the transmit and receive ends of the analog or digital loopback path.)

2.7.1.4 ESD/ESCD Selection Requirement (MER)

The STU-III shall contain an ESD/ESCD "strap". When the strap is in the ESD position, the modem shall transmit the ESD tone in accordance with the

F. Intracharacter Receive Signal Rate: In the receiving LCT, the intracharacter signaling rate output to its DTE shall be in the range of 2400 to 2455 bps.

G. Receive Break Signal: Same as in Section 2.5.2.1.2.

2.5.2.3.3 Black Side Data Rate Transmission Format. The STU-III shall transmit the asynchronous data to the far-end terminal in a 2400 bps synchronous transmission. There shall be no data bits lost in the encryption/decryption process (i.e., the first bit out of the data device shall be the first bit encrypted by the first bit of key, and shall be the first bit transmitted after START; there will be no sacrifice bits in the data mode caused by the encryption process). The black side of the terminal shall not incorporate the BCH code while in this mode.

2.6 COMSEC

ALL OF SECTION 2.6 IS PROVIDED IN A SEPARATE CLASSIFIED ATTACHMENT

2.7 MODEM PROCESSING

This section defines the requirements for the STU-III modems. The STU-III shall include the 2400 bps modem specified in Section 2.7.1 to support interoperability with all vendors in the full duplex and half duplex modes of operation. In addition, the STU-III may contain modems for operation at 4800 and 9600 bps as specified in Section 2.7.2. The requirements for modulation, message scrambling, and coding provisions for these modes are defined in Section 3 of FSVS-210.

2.7.1 2400 bps Modem (MER)

The STU-III/LCT shall incorporate a 2400 bps modem with the modulation scheme defined by CCITT recommendation V.26 using B modulation with echo

A. Signal-to-Noise Ratio: The signal to noise ratio (SNR) should be 50 dB minimum between the modem and the telephone line (transmit or receive path) when measured from 300 Hz to 3 KHz bandwidth at rated levels at the telephone line (0 dBm transmit path and -15 dBm receive path at 1800 Hz).

B. Frequency Response: The frequency response should be as follows:

(1) 50 Hz - Down 20 dB minimum
(2) 100 Hz - Down 13 dB maximum
(3) In Band (270 Hz to 2700 Hz) - ±.5 dB ripple
(4) 3240 Hz - Down 4 dB maximum
(5) 3600 Hz - Down 16 dB minimum
(6) Above 4000 Hz - Down 30 dB minimum
(7) Above 4500 Hz - down 35 dB minimum.

C. Total Distortion: The total distortion should be at least 40 dB below a 300 Hz signal level between the modem output and the telephone line for rated levels (0 dBm nominal) measured at the telephone line, and at least 40 dB below a 300 Hz signal level between the telephone line and the modem receiver at -15 dBm/600 ohms at the telephone line input.

D. Transmit Signal Level: The transmit signal levels should be related as follows before any transmit path gain adjustment:

(1) Modem signal (1800 Hz)
0 dBm/600 ohms ±1 dBm

(2) DTMF Levels
Combined: 0 dBm/600 ohms ±1 dBm
High band: -2 dBm/600 ohms
Low band: -3 dBm/600 ohms

cancelling for 2-wire full duplex communication. The same modem shall be used without echo cancelling for operation in half duplex. The terminal modem shall provide the capability for the transmission and reception of digital information over a variety of analog telecommunications channels. This capability shall be demonstrated by operating the modem over the channels specified by Section 2.7.1.1 while meeting the specific performance requirements specified by Section 2.7.1.2. These performance requirements shall be met by the 2400 bps modem in both full and half duplex operation.

2.7.1.1 Channel Characteristics for V.26 Modem Testing

The STU-III shall operate on signals received over a variety of specified transmission channels. Spectral shape and delay characteristics are specified below for eight wireline channels, along with associated impairment parameters. In addition, the specifications below define the impairments to be included when the performance requirements call for channels with near-end and far-end echo, and transmission over ADPCM and cellular radio links.

A. Government Simulated Wireline Channels (MER): The Government has identified eight wireline channels that are representative of the telephone circuits a CONUS-based terminal/user might expect to encounter. Tables 2-2 and 2-3 and Figures 2-4 and 2-5 characterize the eight Government simulated wireline channels. When identified in a performance requirement, the specified wireline channels shall be used when modem performance is measured.

B. Adaptive Differential Pulse Code Modulation (ADPCM): The performance requirements may call for impairments due to transmission links using Adaptive Differential Pulse Code Modulation (ADPCM). When impairments due to ADPCM are specified, the simulated channels shall contain impairments that represent four hops of 32 kbps mu-law ADPCM in addition to the other characteristics or impairments specified.

D. Cellular Radio Operations (MER): The STU-III modem shall retain system synchronization during signal dropouts and fades specified for the cellular medium. The terminal shall bridge modem baud sync when the channel signal fades or drops out. If a fade or dropout occurs, the terminals at each end shall detect the loss of carrier, freeze the equalizer parameters for the echo cancellers and modem adaptive equalizers, squelch the audio output and maintain timing to permit operation immediately without crypto or modem resynchronization when the channel signal returns. It is desirable for the terminals to verify that the voice processor is still in frame sync following a fade before the synthesizer output is enabled to the handset.

The STU-III modem shall retain system synchronization during the specified instantaneous change in the transmission path delay and shall continue processing data at the conclusion of the change without resynchronization.

After either a fade, dropout or instantaneous change in the transmission path delay, the modem shall restore reception to the specified BER performance level within 100 milliseconds, assuming no change in line impairments.

The cellular radio operational requirements shall be met by demonstrating compliance on the CONUS Mid Data (CMD) line without impairments due to ADPCM and echo.

2.7.1.3 Telephone Line/Modem Interface Recommendations

The following telephone line/modem interface specifications are recommended to provide reliable modem performance. The telephone line/modem interface should consider the analog voice requirements between the telephone line and the modem, the DTMF generator and the plaintext circuits.

Table 2-2. Channel Simulation Test Characteristics

Channel Simulated	Signal to Noise Ratio	Impulse Hits (per 15 Min)	Phase Jitter	Phase Hits (per 15 Min)	Frequency Offset	Harmonic Distortion
CONUS Poor Voice	27 dB	—	15° @ 60 Hz	45 @ 17°	± 1Hz	- 34 dB
CONUS Mid Voice	33 dB	—	12° @ 60 Hz	15 @ 17°	—	- 39 dB
CONUS Poor Data	24 dB	—	14° @ 60 Hz	45 @ 17°	± 1Hz	- 34 dB
CONUS Mid Data	32 dB	—	11° @ 60 Hz	5 @ 17°	—	- 42 dB
European Poor Voice	26 dB	225	35° @ 50 Hz	405 @ 42°	± 7Hz	- 51 dB
European Mid Voice	32 dB	25	26° @ 50 Hz	45 @ 32°	± 3 Hz	- 57 dB
European Poor Data	26 dB	225	35° @ 50 Hz	135 @ 42°	± 6 Hz	- 51 dB
European Mid Data	32 dB	25	18° @ 50 Hz	135 @ 22°	± 3 Hz	- 57 dB

Notes:
1. Channel simulators may assume Gaussian Noise.
2. Impulse and phase hits specified in the number of hits per 15 minutes.
3. Impulse hits are at 74 dBrnc0 with a duration of 100 microseconds.
4. Phase jitter is specified in degrees peak-to-peak.
5. Phase hits specified in degrees at the peak of the hit. Duration of the hit is 2 milliseconds.
6. Harmonic distortion is specified as levels in dB with respect to the transmit signal level. Values specified apply to both the 2nd and 3rd harmonic distortion.

Table 2-4. Performance Requirements for 2400 bps Modems

Transmission Channel Conditions						Requirements for V.26 Modem Operation at 2400 bps
Government Lines	ADPCM (32 kbps)	Near-End Echo	Far-End Echo	Cellular Impairment	Received Signal Level	
All 8	4 Hops	—	—	—	0 to -43 dBm	-Initial Synchronization ≥ 99% -Retain Synchronization > 10 min -MER Intelligibility and Quality -BER ≤ 5 x 10 $^{-4}$
All 8	4 Hops	E ≤ -12 dBm D ≤ 12 ms	—	—	0 to -43 dBm	-Initial Synchronization ≥ 99% -Retain Synchronization > 10 min -MER Intelligibility and Quality -BER ≤ 5 x 10 $^{-4}$
All 8 (With no frequency offset)	4 Hops	E ≤ -12 dBm D ≤ 12 ms	S/E ≥ -3 dB D ≤ 750 ms	—	-15 to -36 dBm	-Initial Synchronization ≥ 99% -Retain Synchronization > 10 min -MER Intelligibility and Quality -BER ≤ 5 x 10 $^{-4}$ (Except EPV)
CMD	—	—	—	≤ 5 sec fade or dropout	0 to -43 dBm	-Retain System Synchronization -Restore BER ≤ 5 x 10 $^{-4}$ within 100 ms after dropout
CMD	—	—	—	up to ± 270 microsecond instantaneous change in path delay	0 to -43 dBm	-Retain System Synchronization -Restore BER ≤ 5 x 10 $^{-4}$ within 100 ms after change

B. Terminal End-to End Performance (MER): After synchronizing, the STU-III shall achieve MER intelligibility and quality of the LPC-10(e) with another STU-III when tested over all eight Government simulated wireline channels. This requirement shall be met independently on each of the eight Government lines, with ADPCM, and with or without near-end and far-end echo.

C. Bit Error Rate Performance (MER): The STU-III modem shall operate at a bit error rate (BER) of $5x10^{-4}$ or better. This requirement shall be met independently on each of the eight Government lines, with ADPCM, and with or without near-end and far-end echo. When far-end echo is present, however, no BER requirement is specified for the European Poor Voice (EPV) line. Notwithstanding the absence of a requirement, it is recommended that terminals maintain the BER of $5x10^{-4}$ or better on the EPV line.

Table 2-3. Delay and Amplitude Values for 8-Government Simulated Wireline Channels

		CONUS POOR VOICE		CONUS MID VOICE	
N	FREQ (Hz)	AMP (dB)	DELAY (ms)	AMP (dB)	DELAY (ms)
1	0.00	-40.0	4.00	-40.0	4.00
5	156.25	-40.0	4.00	-40.0	4.00
9	312.50	-21.0	4.00	-7.3	4.00
13	468.75	-4.3	4.00	-4.5	3.73
17	625.00	-1.8	3.14	-2.7	2.35
21	781.25	-0.7	2.28	-1.4	1.41
25	937.50	-0.1	1.55	-0.4	.61
29	1093.75	0.0	1.02	+0.5	.38
33	1250.00	-0.4	.63	+1.2	.25
37	1406.25	-1.0	.34	+1.6	.06
41	1562.50	-1.9	.14	+1.6	.01
45	1718.75	-3.8	.04	+0.9	.05
49	1875.00	-5.5	.04	+0.1	.07
53	2031.25	-7.5	.13	-0.6	.09
57	2187.50	-9.8	.31	-1.3	.13
61	2343.75	-12.2	.53	-1.7	.20
65	2500.00	-15.0	.80	-2.0	.33
69	2656.25	-17.9	1.18	-3.1	.67
73	2812.50	-21.2	1.68	-4.8	1.39
77	2968.75	-25.2	2.97	-7.0	2.65
81	3125.00	-29.3	6.37	-10.8	3.63
85	3281.25	-33.3	6.17	-16.7	4.00
89	3437.50	-37.6	5.97	-21.5	4.00
93	3593.75	-39.5	5.78	-26.8	4.00
97	3750.00	-39.6	5.58	-32.3	4.00
101	3906.25	-39.7	5.38	-37.7	4.00
105	4062.50	-39.8	5.18	-40.0	4.00
109	4218.75	-39.9	4.99	-40.0	4.00
113	4375.00	-39.9	4.79	-40.0	4.00
117	4531.25	-40.0	4.59	-40.0	4.00
121	4687.50	-40.0	4.39	-40.0	4.00
125	4843.75	-40.0	4.20	-40.0	4.00

F. Received Signal Level: The V.26 modem shall meet the performance requirements specified below at any received line signal level in the range of 0 dBm to -43 dBm with a recommended range extending to -50 dBm. For testing purposes, this requirement is reduced to received signal levels in the range of -15 dBm to -36 dBm when far-end echo is specified in the channel.

2.7.1.2 Performance Requirements for V.26 Modems

While operating on signals received over a variety of transmission channels, the STU-III operating with the V.26 modem shall meet performance requirements on initial and long term system synchronization, voice intelligibility and quality, bit error rates, and recovery after impairments in a cellular environment. Table 2-4 specifies the channel characteristics and identifies the requirements that must be met when operating over the specified channel. Further details of the channel characteristics identified in Table 2-4 are specified in Section 2.7.1.1 while details of the performance requirements are specified below. These performance requirements shall be met by the STU-III modem in both the full and half duplex modes of operation.

A. Synchronization (MER): After a secure call is initiated, the STU-III modem shall acquire system synchronization on at least 99 percent of the secure call setup attempts without relying on the retransmission protocol specified in FSVS-210, Section 2.2.1.3.1. The STU-III modem shall be designed to maintain synchronization 100 percent of the time after acquisition. Demonstration tests to verify the ability to retain synchronization shall be conducted for a minimum of 10 minutes. These synchronization requirements shall be met independently on each of the eight Government lines, with ADPCM, and with or without near-end and far-end echo.

Table 2-3. Delay and Amplitude Values for 8-Government Simulated Wireline Channels (Cont.)

		CONUS POOR DATA		CONUS MID DATA	
N	FREQ (Hz)	AMP (dB)	DELAY (ms)	AMP (dB)	DELAY (ms)
1	0.00	-40.0	4.00	-40.0	4.00
5	156.25	-40.0	4.00	-20.7	4.00
9	312.50	-10.1	3.74	-4.6	3.75
13	468.75	-2.6	.94	-1.4	.39
17	625.00	-1.6	.02	-0.7	-.12
21	781.25	-0.8	.39	-0.1	-.02
25	937.50	-0.2	.61	0.0	.02
29	1093.75	+0.5	.51	+0.2	-.05
33	1250.00	+1.0	.50	+0.9	-.01
37	1406.25	+0.8	.50	+1.0	.00
41	1562.50	+0.2	.52	+0.5	.01
45	1718.75	-0.5	.56	0.0	.05
49	1875.00	-1.4	.62	-0.4	.10
53	2031.25	-2.2	.65	-1.1	.10
57	2187.50	-2.9	.65	-1.5	.10
61	2343.75	-3.5	.58	-1.8	.06
65	2500.00	-4.0	.50	-2.0	-.02
69	2656.25	-4.6	.56	-2.6	-.13
73	2812.50	-6.1	.57	-3.1	-.18
77	2968.75	-7.7	.53	-4.2	-.03
81	3125.00	-11.8	.86	-14.6	.28
85	3281.25	-21.7	1.85	-40.0	.83
89	3437.50	-32.5	3.38	-40.0	1.18
93	3593.75	-40.0	4.00	-40.0	1.49
97	3750.00	-40.0	4.00	-40.0	1.80
101	3906.25	-40.0	4.00	-40.0	2.11
105	4062.50	-40.0	4.00	-40.0	2.51
109	4218.75	-40.0	4.00	-40.0	3.04
113	4375.00	-40.0	4.00	-40.0	3.58
117	4531.25	-40.0	4.00	-40.0	4.00
121	4687.50	-40.0	4.00	-40.0	4.00
125	4843.75	-40.0	4.00	-40.0	4.00

C. Near-End Echo: When specified by the performance requirements, impairments due to near-end echo shall be added to the received line signal. For testing purposes, the near-end echo shall be a single linear reflection of the transmitted line signal. The echo shall occur at levels of up to -12 dBm and delays of up to 12 milliseconds.

D. Far-End Echo: When specified by the performance requirements, impairments due to far-end echo shall be included in the received line signal. For testing purposes, the far-end echo shall be a single linear reflection at the far end of the transmitted line signal. The far-end echo shall be added to the signal transmitted by the far-end terminal and the resulting signal shall be subjected to all of the impairments defined for the specified channel. However, for testing purposes, impairments due to frequency offset shall not be required in the channel when far-end echo is specified. When far-end echo is specified, the received signal to far-end echo (S/E) ratio shall be greater than or equal to -3 dB.

For testing purposes, the far-end echo shall be present in the received line signal at delays of up to 750 ms. Notwithstanding this requirement, however, it is recommended that terminals be designed to operate with far-end echoes delayed by up to 1250 ms.

E. Cellular Radio Impairments: When specified by the performance requirements, impairments typical of cellular radio links shall be inserted in the channel in addition to other characteristics specified. For testing purposes the cellular radio impairment shall include either a signal fade, or an instantaneous change in the transmission path delay. The signal fade shall be either a fade or a complete signal dropout lasting up to 5 seconds. The instantaneous change in transmission path delay shall be up to ± 270 microseconds (± 50 miles).

Table 2-3. Delay and Amplitude Values for 8-Government Simulated Wireline Channels (Cont.)

		EUROPEAN POOR VOICE		EUROPEAN MID VOICE	
N	FREQ (Hz)	AMP (dB)	DELAY (ms)	AMP (dB)	DELAY (ms)
1	0.00	-40.0	4.00	-40.0	4.00
5	156.25	-40.0	4.00	-40.0	4.00
9	312.50	-12.5	4.00	-14.0	4.00
13	468.75	-5.0	2.45	-4.0	2.15
17	625.00	-2.2	1.85	-1.5	1.20
21	781.25	-1.0	1.35	-1.2	0.70
25	937.50	-0.2	0.80	-0.2	0.40
29	1093.75	-0.2	.50	+0.2	0.22
33	1250.00	-0.9	.30	+0.5	0.10
37	1406.25	-1.6	0.00	+1.0	-0.05
41	1562.50	-1.8	-0.20	+1.2	-0.05
45	1718.75	-1.9	-0.30	+1.1	0.00
49	1875.00	-1.3	-0.35	+1.3	0.02
53	2031.25	-0.9	-0.40	+1.3	0.01
57	2187.50	-1.0	-0.45	+1.1	-0.01
61	2343.75	-2.0	-0.35	+1.1	-0.01
65	2500.00	-4.0	-0.25	+1.0	0.05
69	2656.25	-6.9	-0.10	+1.0	0.15
73	2812.50	-7.9	.30	+1.0	0.30
77	2968.75	-6.3	.70	+0.9	0.45
81	3125.00	-4.1	1.10	+0.8	0.95
85	3281.25	-3.6	1.80	+0.1	2.75
89	3437.50	-5.0	3.38	-2.4	4.00
93	3593.75	-13.7	4.00	-40.0	4.00
97	3750.00	-40.0	4.00	-40.0	4.00
101	3906.25	-40.0	4.00	-40.0	4.00
105	4062.50	-40.0	4.00	-40.0	4.00
109	4218.75	-40.0	4.00	-40.0	4.00
113	4375.00	-40.0	4.00	-40.0	4.00
117	4531.25	-40.0	4.00	-40.0	4.00
121	4687.50	-40.0	4.00	-40.0	4.00
125	4843.75	-40.0	4.00	-40.0	4.00

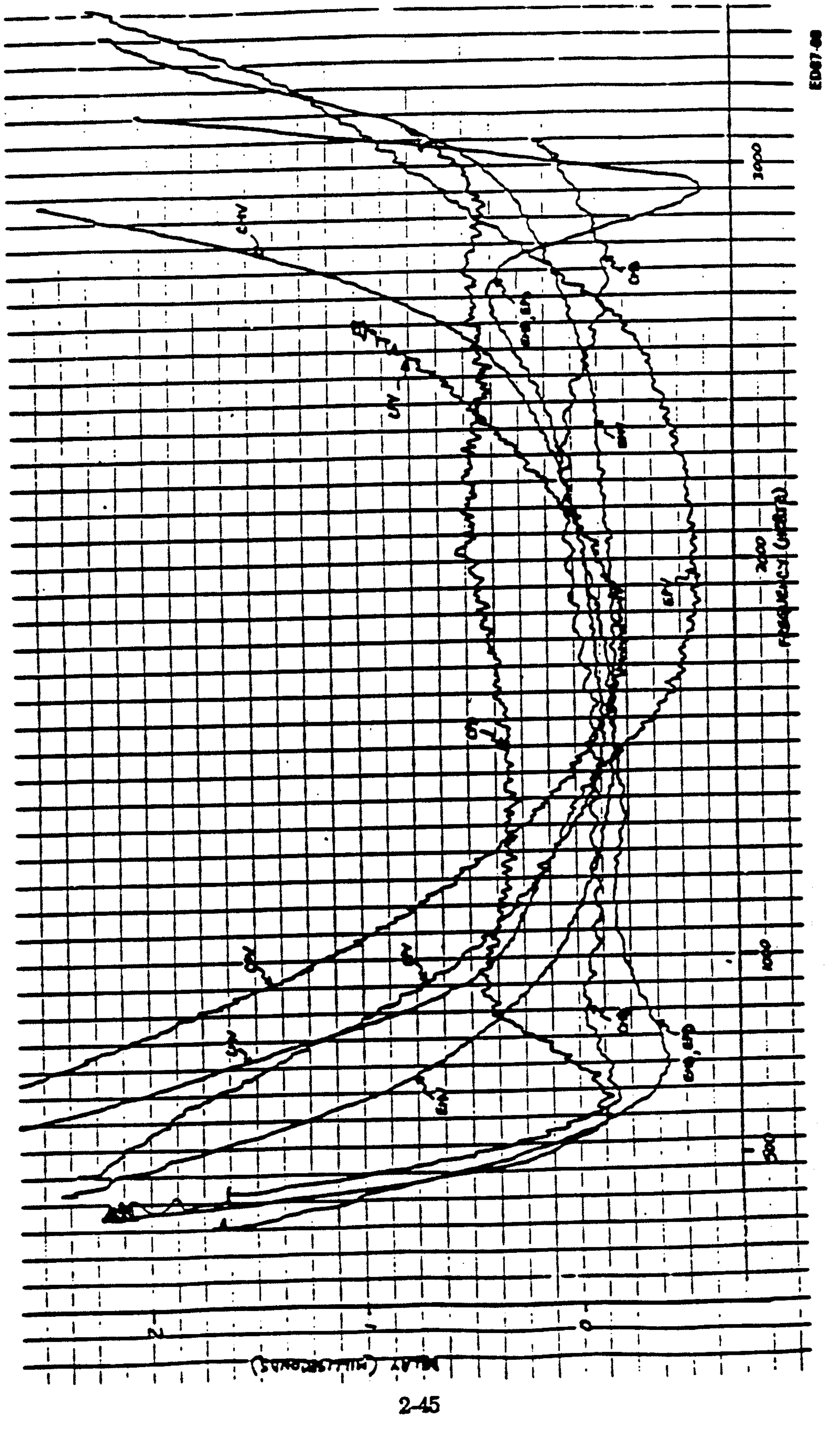

Figure 2-5. Wireline Simulator Delay Responses

Table 2-3. Delay and Amplitude Values for 8-Government Simulated Wireline Channels (Cont.)

		EUROPEAN POOR DATA		EUROPEAN MID DATA	
N	FREQ (Hz)	AMP (dB)	DELAY (ms)	AMP (dB)	DELAY (ms)
1	0.00	-40.0	4.00		
5	156.25	-40.0	4.00		
9	312.50	-11.5	2.35		
13	468.75	-6.4	.65		
17	625.00	-3.5	-.14		
21	781.25	-1.6	-.24		
25	937.50	-0.3	-.10		
29	1093.75	+0.4	-.01	SAME	AS
33	1250.00	+0.8	-.02		
37	1406.25	+1.0	-.03		
41	1562.50	+1.0	.03	EUROPEAN	
45	1718.75	+1.0	.05		
49	1875.00	+0.7	.09		
53	2031.25	+0.3	.13	POOR	DATA
57	2187.50	-0.2	.19		
61	2343.75	-0.9	.21		
65	2500.00	-1.6	.30		
69	2656.25	-2.2	.46		
73	2812.50	-2.8	.37		
77	2968.75	-2.7	-.57		
81	3125.00	-1.4	1.48		
85	3281.25	-0.8	3.63		
89	3437.50	-3.5	4.00		
93	3593.75	-11.6	4.00		
97	3750.00	-21.5	4.00		
101	3906.25	-30.5	4.00		
105	4062.50	-38.3	4.00		
109	4218.75	-38.9	4.00		
113	4375.00	-39.5	4.00		
117	4531.25	-40.0	4.00		
121	4687.50	-40.0	4.00		
125	4843.75	-40.0	4.00		

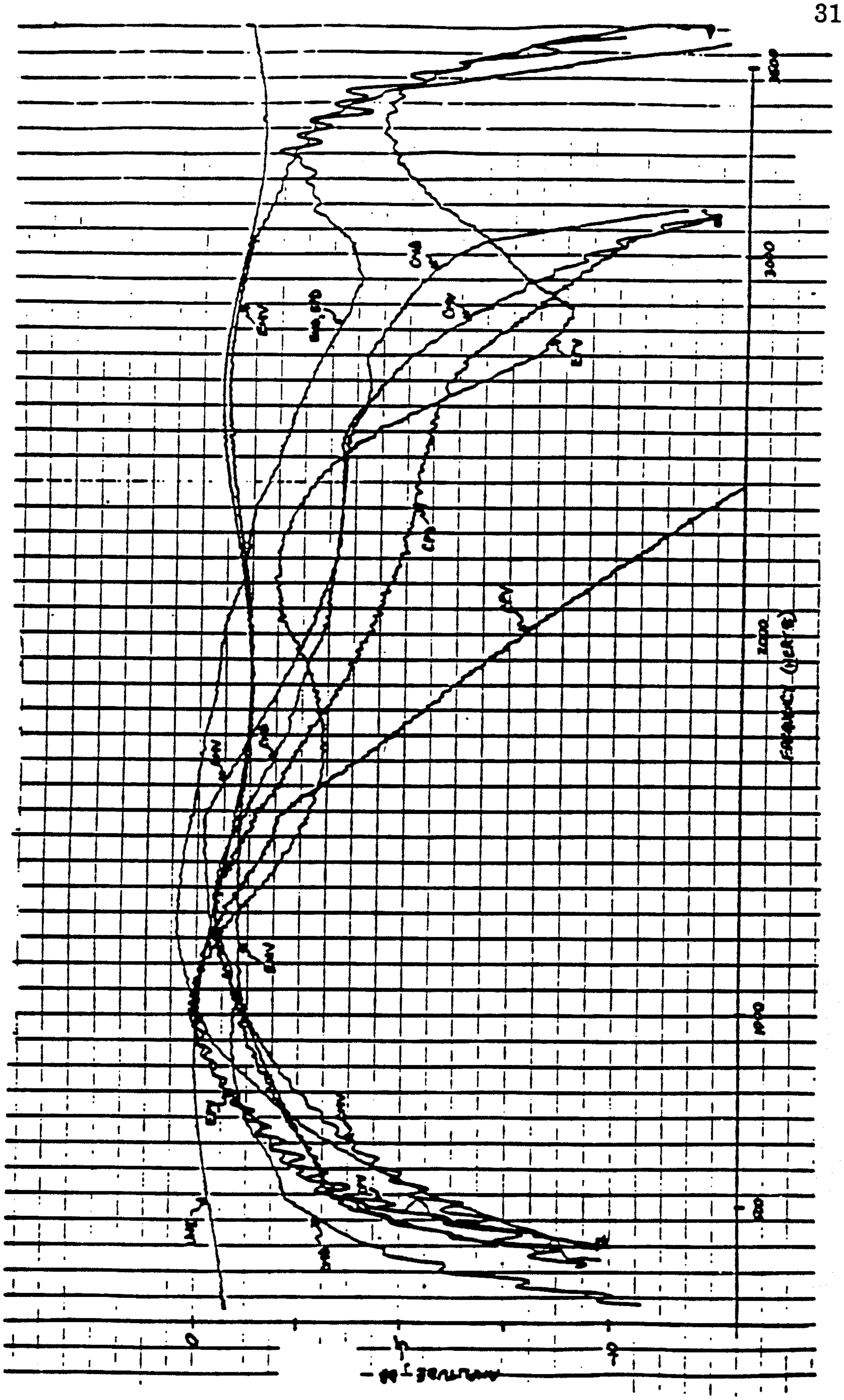

Figure 2-4. Wireline Simulator Amplitude Responses